AN INVESTIGATION OF DIGITAL SPECTRAL ANALYSIS PROGRAMS AND COMPUTER METHODS UTILIZED AT THE NAVAL POSTGRADUATE SCHOOL IN THE ANALYSIS OF HIGH FREQUENCY RANDOM SIGNALS

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THESIS

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March 1972

An Investigation of Digital Spectral Analysis Programs and Computer Methods Utilized at the Naval Postgraduate School in the Analysis of High Frequency Random Signals

by

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ABSTRACT

The digitizing procedure used at the Naval Postgraduate School was investigated for possible sources of noise and other errors. Signals of known form were digitized through the Analog-to Digital Hybrid computer system (Ci 5000/XDS9300). Similar signals were generated by digital programs on the IBM 360/67. The resultant signals were analyzed by the computer programs UBCFTOR, which computed the Fourier coefficients of each block of data, and by UBCSCOR, which computed the power spectra of the signals. The power-spectral plots of the computer-generated signals were compared with the power-spectral plots of digitized signals. The analog-to-digitital process appeared to be relatively noise free.

To further test the system, atmospheric temperature and wind velocity signals were digitized and analyzed under UBCFTOR and UBCSCOR. Plots of the time-varying spectra of these signals compared favorable with results obtained at other digitizing facilities.

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The author would like to acknowledge the invaluable advice on all phases of this research offered by Dr. Noel E. J. Boston. Special thanks are offered to Mr. Robert Limes for advice on the Hybrid Computer System, and Miss Sharon Raney for assistance in programming and in IBM tape utilization procedures. Finally, I wish to express appreciation to my wife for her patience with long hours of typing and checking the manuscript.

I. INTRODUCTION

A. PROBLEM

Random processes are often recorded in a fluctuating voltage in analog form. However, it is frequently more convenient to analyze the signal digitally (on large digital computers). Thus, investigators are faced with the problem of converting a continuous signal into discrete data samples, and with subsequent analysis of these digitized samples. There are several steps in the analog-to-digital conversion procedure, and in the digital analysis procedure, which allow for errors and for possible contamination of a signal with noise from external sources.

B. OBJECTIVE

The main goal of this study was to investigate digitization and analysis procedures for possible sources of noise which may be introduced to the true signals. The overall procedure used at the Naval Postgraduate School, from digitaization of the actual signal to the computation of the Power-Spectral-Density (PSD), is quite complex and requires four computer programs. The next objective was to improve the routine procedures required in this particular time series analysis technique. The final objective was to digitize actual geophysical signals and compare the results with similar analyses of these signals undertaken at the University of British Columbia, by Boston in 1970 [Ref. 1].

II. THEORY

A. DIGITAL REPRESENTATION OF CONTINUOUS, TIME-VARYING SIGNAL

Random geophysical processes are often studied by recording a continually changing event as a continuous, fluctuating voltage, which corresponds linearly with the original process. The actual geophysical variables considered later in this study were small-scale fluctuations of air temperature, wind velocity, and time derivatives of both.

1. Analog-to-Digital Conversion

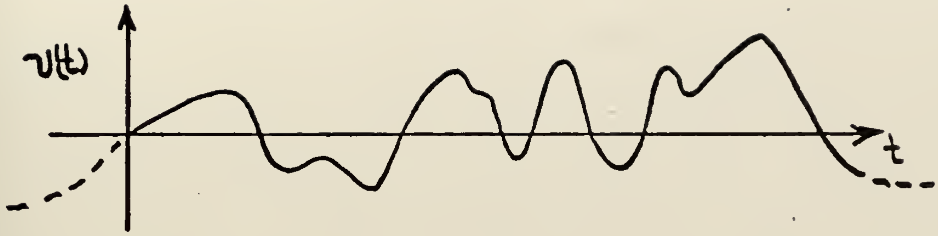
A randomly fluctuating voltage signal might look like the one in Figure 1(a). In order to analyze the signal by digital techniques, discrete samples of the fluctuating voltage must be taken, and these are referred to as sequential digital samples (vi). The requirement for sampling at equal intervals of time is set by the assumption, in most analyses, that they are equal time interval samples. The digitized samples would look like Figure 1(b).

2. Digital Sampling Theory

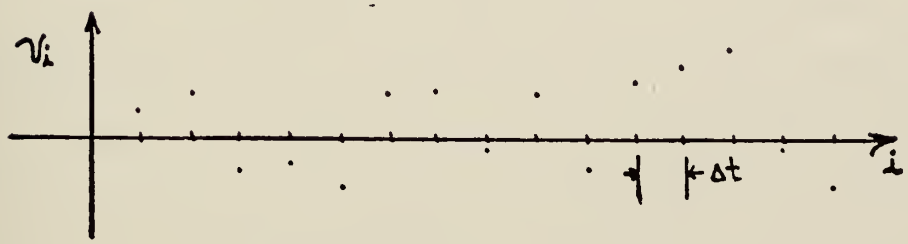
Implied, in the digitization problem, is the question of how well the sequence represents the original voltage. In turning to Sampling Theory for the answer, three hypotheses are made:

a. $V(t)$ is a random variable defined for $-\infty < t < \infty$.

b. The power spectral density of $G(f)$ exists.



a) Analog Signal



b) Digital representation of analog signal

Figure 1. Analog and Digital Signals

c. $G(f)$ equals zero for frequencies equal to, or greater than B. B is the highest frequecy which can be resolved and f is defined as frequency.

Further, if we let Δt be the sampling interval in seconds, so that $\Delta t \leq 1/(2B)$ and if we have $v(t)$ sampled at intervals of Δt, giving vi samples, $i = -\infty, \ldots, 0, 1, \ldots, \infty$, it can be shown that $v(t)$ can be reconstructed uniquely. So, if the power in $v(t)$ is limited to a band less than $1/(2\Delta t)$Hz, then, sampling at an interval Δt allows $v(t)$ to be reconstructed uniquely. Proof of this augument is given in [Ref. 2]. In this way, the requirement of sampling at least twice per cycle has not only been established but is entirely sufficient.

3. Limitations on Frequency Resolution

Due to real world limitations of not being able to collect a signal $v(t)$ of infinite length, and because our actual signals are not necessarily band-limited, we have to modify many of our theoretical assumptions. Firstly, we assume we can get a long record length, which is representative, at least over the range of frequencies we are interested in, of the signal extending in time to plus and minus infinity; secondly, we can use "low-pass" filters to eliminate unwanted high frequencies before the signal is sampled.

a. Low Frequency Limitations

If the signal $V(t)$ contains low frequencies it will be very hard to distinquish then in the interval $0 \leq f \leq 1/(2T)$, according to Rayleigh's Criterion, [Ref. 2], where T is the record length in seconds. In this situation, when recording

a finite section of signal, we have, in fact, truncated the original signal. The recorded section of data now represents the original signal. The term "block" refers to a further truncation of the signal into smaller sequential sections of data, which, when added sequentially, will give a truncated, sampled, section of signal. Thus as T increases, or the length of signal examined increases, the lower will be the frequency which can be resolved.

b. High Frequency Limitations

The highest frequency we can resolve has been established as one-half the sampling rate. In other words, at least two samples per cycle are required. The high frequency limit is more commonly known as the "high frequency cut-off point," or just "cut-off frequency," (f_c). Sometimes, requirements of five samples per cycle are set; however, this added computational problem is in fact unnecessary and essentially means that the high frequency limit of analysis is increased by a factor of 2.5. If 1000 Samp/Sec were required for a high frequency limit of 500 cycles, 2500 Samp/Sec would result in a high frequency limit of 1250 cycles. Usual sampling rates vary between 2 and 2.5 samples per cycle. In this study, a sampling rate of two samples per cycle was used for the turbulence analysis. This rate had been established as satisfactory by Boston (1970), [Ref. 1].

c. Aliasing

The requirement of simply sampling the highest frequency of interest at two samples per cycle is adequate if the highest frequency of interest is, in fact, the

highest frequency present in the signal. When higher frequencies are present, the cut-off frequency becomes the "folding" or "Nyquist" frequency. The "folding" comes from the fact that higher frequency energy, "electrical energy" in our case, is "folded" back into lower frequencies around this point. To eliminate the problem it is necessary to either sample at higher rates, thus moving the folding frequency higher, or to sharply filter the signal at the folding frequency.

B. FOURIER TRANSFORMATION

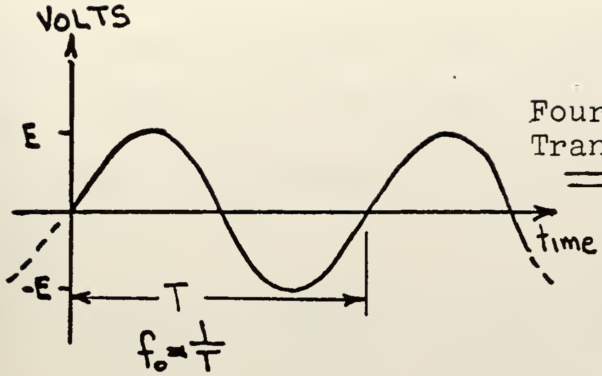
Many studies of random geophysical processes seek to describe the distribution of energy within a varying process as a function of the frequency of fluctuation of the variable being measured. Sometimes when analysing turbulence, it is desired to determine the distributation of energy in the rapidly fluctuating velocity as a function of frequency. The problem is, thus, basically one of transforming data from the time-domain into the frequency-domain.

1. Fourier Transformation of a Continuous Signal: Fourier Integral

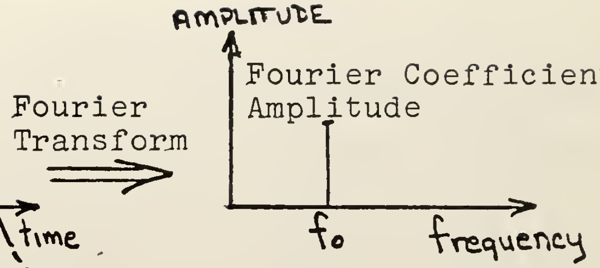
One of the most often used techniques for this transformation, from the time-domain to the frequency-domain, is through the use of the Fourier Intergral Transform. According to this procedure, a periodic sinusoidal signal (shown in analog form in Figure 2a) is transformed from the time-domain into the frequency-domain through the transform-function

$$V(f) = \int_{-\infty}^{\infty} v(t) e^{-j2\pi ft} dt \qquad (1)$$

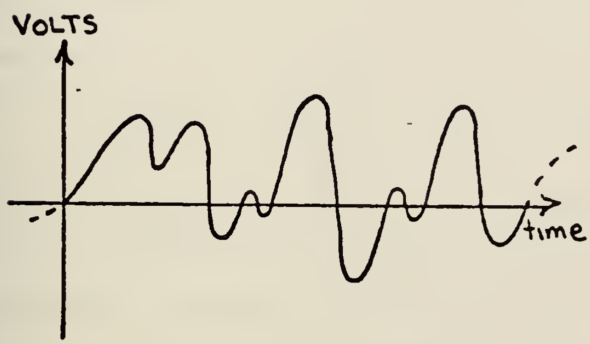
where $v(t) = E \sin 2\pi f_0 t$ and $j = \sqrt{-1}$



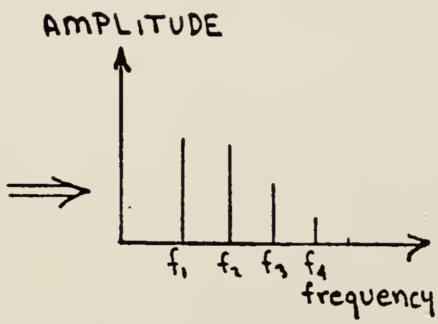
a) Periodic Sinosoidal Signal



b) Fourier coefficient of transformed signal



c) Complex periodic Signal



d) Fourier coefficients of transformed signal

Figure 2. Fourier Transformations

This transformation is shown graphically in Figure 2b). A complex, almost periodic signal composed of several sine waves of differing amplitudes and differing frequencies would be similarly transformed into the frequency-domain, as shown in Figure 2d).

2. <u>Fourier Transform of Discrete Data Signal</u>

When dealing with digitized signals or discrete data the finite form of the Fourier-Transform must be used:

$$V(f_K) = \Delta t \sum_{i=0}^{N-1} v_i \, e^{-j2\pi f_i \Delta t} \qquad -\infty < f < \infty \tag{2}$$

where v is a complex variable and K=0,...,N/2. For real data vi where i=0,...,N-1, the sine and cosine transforms becomes:

$$a(f_K) = \Delta t \sum_{i=0}^{N-1} v_i \text{ Sin } 2\pi f_i \Delta t \tag{3a}$$

$$b(f_K) = \Delta t \sum_{i=0}^{N-1} v_i \text{ Cos } 2\pi f_i \Delta t \tag{3b}$$

Reference 2 suggests that, computationally, this would follow a flow diagram as in Figure 3. The value of f is computed as a function of the number of data points and the length of the record in seconds (T). In other words, increasing the record length decreases the frequency difference between coefficients; or a longer record will give spectral estimates closer together in frequency. Making the substitution for f_K, the sine transformation becomes:

$$a(K\Delta f) = \sum_{i=0}^{N-1} v_i \text{ Sin } 2\pi \frac{iK}{N} \tag{4}$$

This approach to computing Fourier coefficients has been limited in the past, due to the computational time required.

$$f_k = k\Delta f = \frac{K}{T} = \frac{K}{N\Delta t} \quad , K = 0, 1, \ldots N/2$$

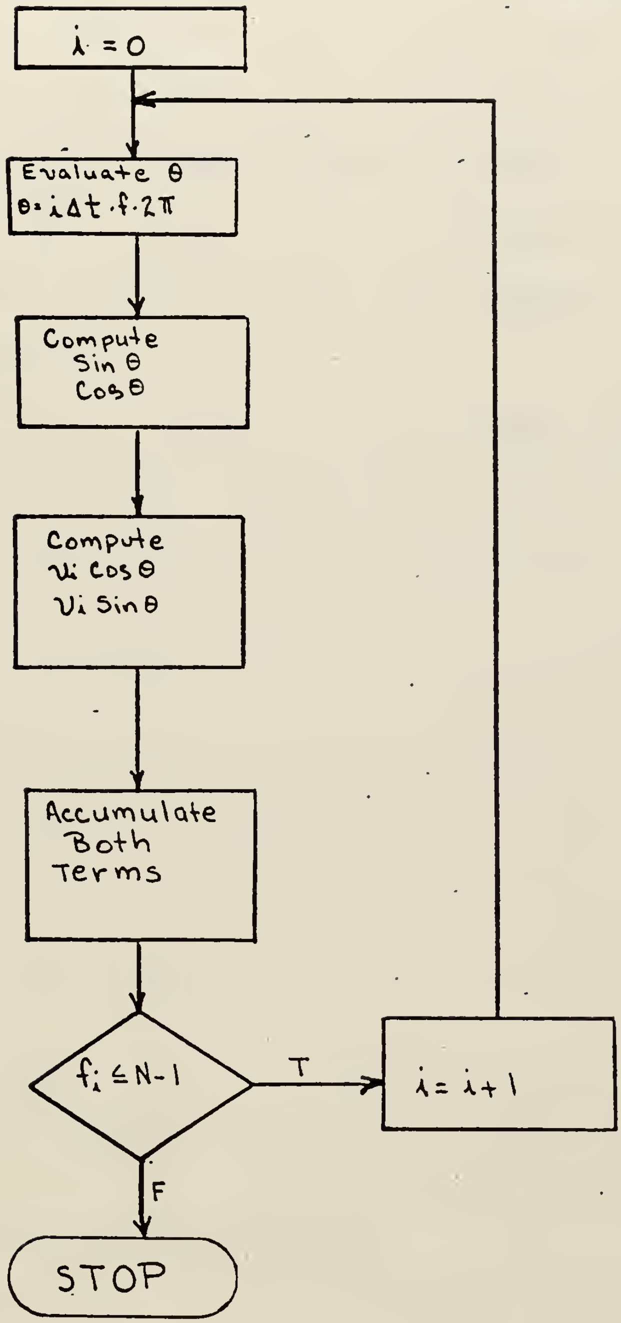


Figure 3. Flow Chart Showing Fourier Transform Procedure for use with Discrete Data

The number of calculations required increases as the square of the number of data points increases. The integral transform, in the past, was limited to studying theoretical functions, such as sine waves and square waves, which were relatively well-behaved functions. For cases such as these, even the discrete transform could be used to readily obtain the Fourier-transform of a signal. With the introduction of the Fast Fourier Transform (FFT), algorithm, computation time has been greatly reduced.

3. Fourier Transform of Truncated Continuous wave Form

If a signal v(t) exists only for the time interval from 0 to T seconds, and is zero at all other times, its Fourier-transform is:

$$V(f) = \int_0^T v(t) e^{-j2\pi ft} dt \tag{5}$$

If v(t) is repeated in intervals of period T seconds, the frequency difference between coefficients will be 1/T Hz. The Kth coefficient will be:

$$V_K = \frac{1}{T} \int_0^T v(t)\, e^{-j2\pi \frac{K}{T} t} dt \tag{6a}$$

and for the discrete case:

$$V_K = \frac{1}{T} \sum_{i=1}^{N/2} v_i\, e^{-j2\pi \frac{K}{T} i} \tag{6b}$$

4. Convolution of Continuous Signals

The finite-transform of a finite-length time series can be viewed as the product of a finite-length rectangular function $C_{T/2}(t)$, times an infinitely long-time history v(t). The finite transform of v(t) becomes:

$$V(f) = \int_0^T C_{T/2}(t)\, v(t)\, e^{-j2\pi ft} dt \tag{7}$$

Since products transorm into convolutions, the convoluted Fourier-transform becomes

$$V_C(f) = V(f) \times \mathcal{C}_{T/2}(f) \tag{8}$$

where

$$\mathcal{C}_{T/2} = \int_0^T C_{T/2}(t)\, e^{-j2\pi ft}\, dt \tag{9}$$

Figure 4(c) shows the Fourier-transformation of the rectangular function and Figure 4(d) shows the convolution of a sine wave with the rectangular function. This convolution problem arises when switches are opened and closed while recording the data. The side lobes of the convolved function can be minimized by allowing the time length of the record to be sufficiently large. This decreases the interval 0 to 1/T.

C. POWER SPECTRAL DENSITY FUNCTION

1. Methods of Computing Power Spectral Density

There are three methods which may be used to compute the power spectral density of a signal. They are:

a. Direct Fourier Transform Method

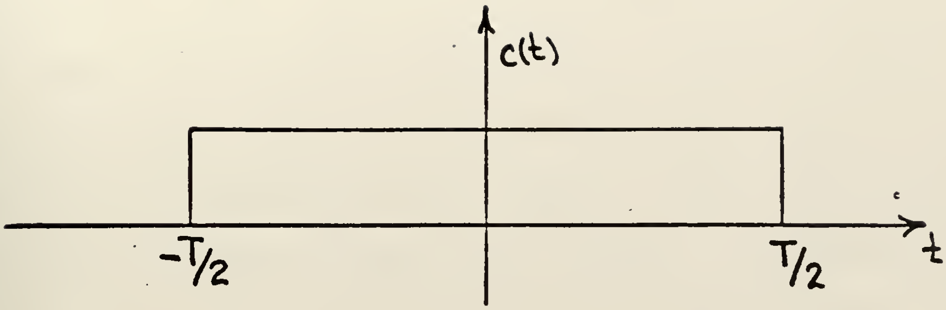
The Fourier transform of the signal is computed and from this the mean value squared is determined.

b. Analog or Bandpass Filter Method

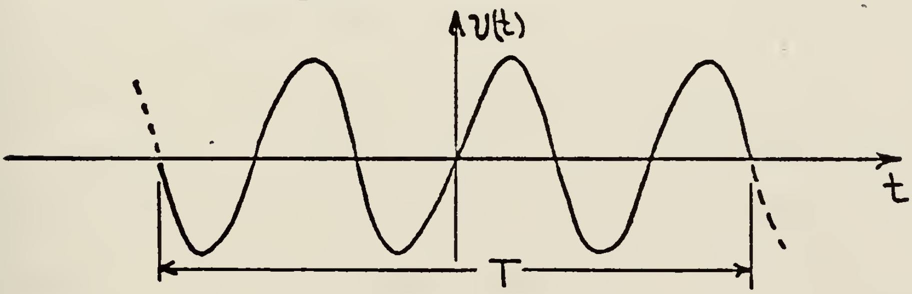
The signal is put through a bank of bandpass filters and each filter output is squared and integrated.

c. Auto-correlation Function Method

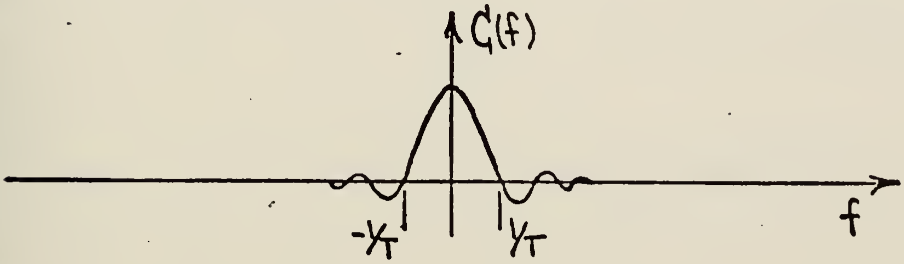
The Auto-correlation function of the time series is computed, and then its Fourier-transform is computed.



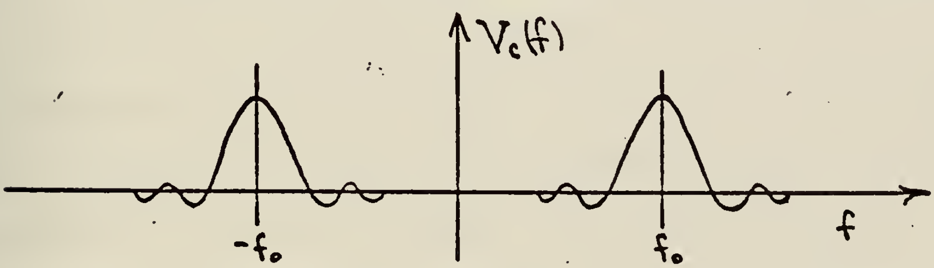
a) Rectangular function



b) Truncated sine function



c) Fourier transform of rectangular function



d) Fourier transform of truncated sine function

Figure 4. Effect of Truncation of Sine Function

Historically, the direct method was developed first, but could be used only on nicely behaved theoretical signals. When less well behaved signals where analyzed, smooth (deterministic) theoretical functions were replaced by discrete data points representing the signal. This necessitated using the discrete Fourier-transform; however, for large data-sets, the time for calculating the Fourier-transform was prohibitive. The computer program FTOR utilized a variation of the direct method employing the Fast Fourier Transform (FFT).

Using the sampling rates 1000-5000 samp/sec, (SPS), and having the ability to select the block sample length and the number of blocks desired for a particular run, no problems were encountered in which filler data, usually zeros, had to be inserted into a block. The block size was always chosen as an integral power of two.

2. Typical Power Spectral Density Functions

The Power Spectral Density(PSD) plot for random data shows the distribution of electrical power within the signal as a function of frequency. Several characteristic PSD plots are encountered. Figure 5(a) is a PSD plot of a pure sine wave. It gives the Dirac-delta function which implies that the power at the sine frequency is infinitely large, and zero at all other frequencies. Figure 5(a) is a PSD of a Gaussian random signal. The PSD of this signal is constant. Figure 5(c) shows the PSD of a random signal carrying a sine function on it. The PSD plot is the sum of the power spectra of the random signal and sine figured separately.

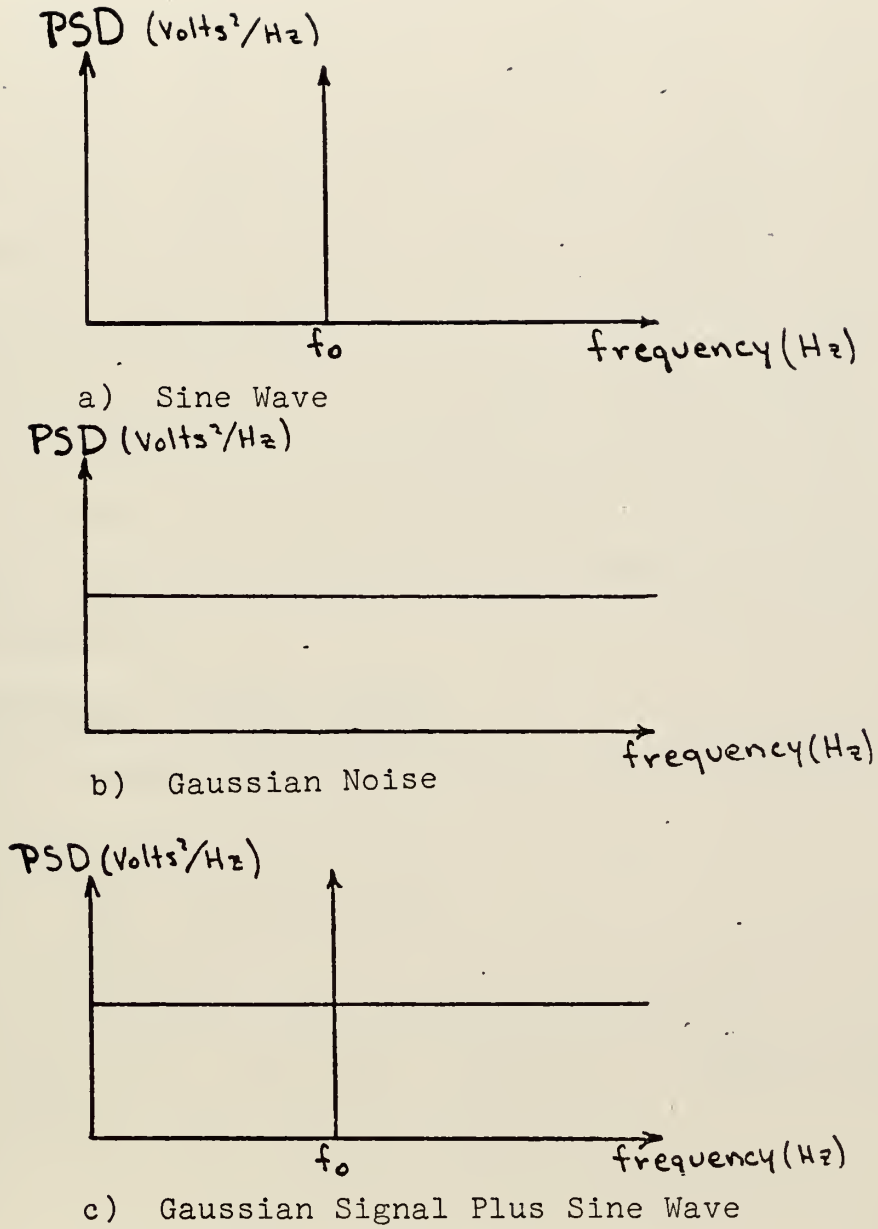


Figure 5. Characteristic PSD Plots

3. Problem of Single-Sided Spectrum

The Fourier coefficients under the transformation

$$Y(f) = \int_{-\infty}^{\infty} y(t)e^{-j2\pi ft}\, dt \qquad (10)$$

exist for both positive and negative frequencies. The coefficients can be transferred to a single-sided frequency plot by simply doubling each coefficient as it is plotted only along the positive frequency axis. If S(f) represents the two-sided spectral density function and G(f) represents the single-sided spectral density function, the single-sided spectral density function equals twice the double-sided density function: G(f)=2S(f). Likewise, if a watt meter was used to find the power in a signal as a function of frequency, the values obtained would have to be divided by two before plotting on a two-sided spectrum.

4. Power and Energy Signals

The average electrical power in a fluctuating voltage signal v(t) is given by

$$P = \lim_{a\to\infty} \frac{1}{2a}\int_{-a}^{a} |v(t)|^2\, dt \qquad (11)$$

Mix [Ref. 3] defined a power signal as one which is, for all practical purposes, infinitely continuous. Energy signals are pulse-like in form and are given by:

$$E = \int_{-\infty}^{\infty} |v(t)|^2\, dt \qquad (12)$$

The power in a periodic signal is shown from Parseval's Theorem to be:

$$P = \frac{1}{T}\int_{t'}^{t'+T} |v(t)|^2 \, dt \qquad (13)$$

Thus, the power in a continuous sine function, v(t)=sin 377t, would be

$$P = \frac{1}{T}\int_0^T \text{Sin}^2 \, 377t \, dt = 1/2 \text{ watt} \qquad (14)$$

If a real wide band signal were to be analyzed by using a tunable filter of band width of δf, a plot of power versus frequency as shown in Figure 6 might result. Here, the real power from zero to an upper frequency was determined, divided by two and the values folded back into negative frequencies.

5. <u>Power from the Fourier Coefficients</u>

Parseval's Theorem leads to the definition of power in terms of the Fourier coefficients

$$P = \frac{1}{T}\int_{t'}^{t'+T} |v(t)|^2 \, dt = \sum_{K=-\infty}^{\infty} |V_K|^2 \qquad (15)$$

Thus, knowing v(t), the average power can be computed. The average power in a one volt, 60Hz sine wave is:

$$v(t) = \text{Sin} 2 \, 60t$$

$$P = \frac{1}{T}\int_0^T \text{Sin}^2 \, 377t \, dt = 1/2 \text{ watts} \qquad (16)$$

Using the Fourier coefficients:

$$V_{-1} = 1/2 \qquad V_1 = 1/2 \qquad V_{K>1} = 0$$

$$P = \sum_{K=-\infty}^{\infty} |V_K|^2 = 1/4 + 1/4 = 1/2 \qquad (17)$$

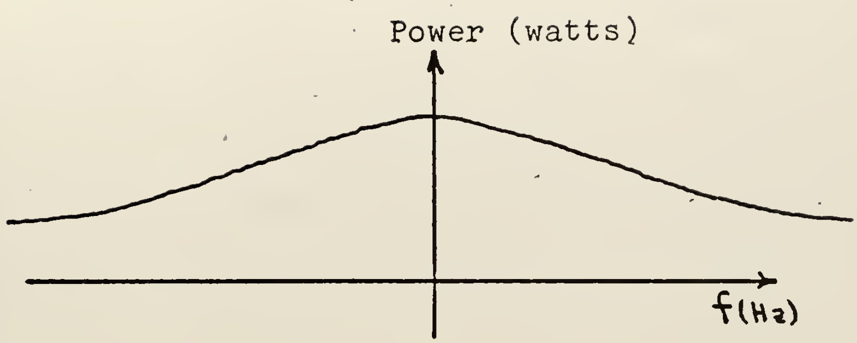


Figure 6. Two-Sided Power Spectrum of Wide Band Signal

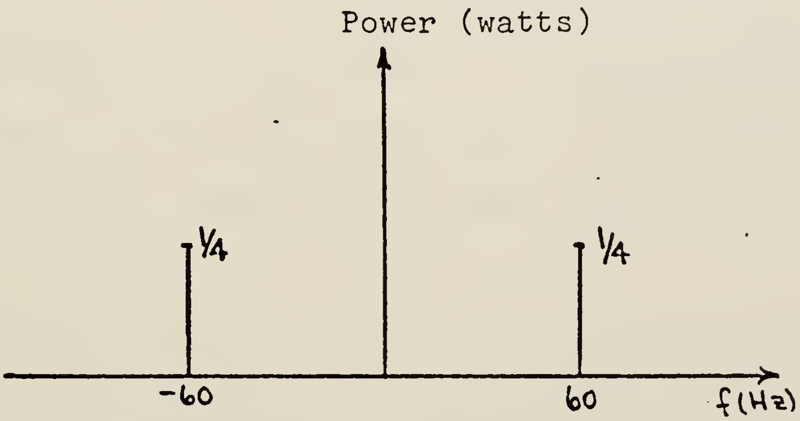


Figure 7. Power Spectrum of 1 volt 60 Hz Sine Signal

If a watt meter was used to check the power of this sine signal, one-half watt would be found at 60Hz, and zero watts at other frequencies. If a plot of power versus frequency were made, one similar to Figure 7 would result. Since the values have been plotted for positive and negative frequencies, the total power has been reduced by one-half and the values have been plotted.

D. FAST FOURIER TRANSFORM

1. Computational Economy Afforded by FFT

The Fast Fourier Transform (FFT) algorithm is a fast method for computing the finite Fourier transform of a series of N data points. The FFT computation requires $N \log_2 N$ computations, which leads to a substantial saving in computer time over the old method which required N^2 operations. The FFT economy of computational effort is greatest for large values of N.

Reference [4] gives the general computational routine for figuring the FFT. It is pointed out that the FFT is general in that N is not necessarily a power of two; however, by selecting N to be a power of two, further computational savings result. When this requirement is met, the FFT algorithm is essentially a successive doubling operation. Thus for $N=2^j$, only Nj multiply-add operations would be required under the FFT assuming the necessary complex exponential table of values has been computed in advance.

2. Importance of FFT in Turbulence Analysis

Computationally, the FFT is an indispensable aid in the PSD investigation of turbulence. Due to the extremely high sampling rates necessary for studying high wave number processes, enormous amounts of data are collected. A five minute section of a fluctuating temperature signal sampled at a rate of 4000 samples/second (SPS) generated almost a million and a half data samples. Using the old method for computing the Fourier transform for this quantity of data, several thousand billion operations would be required. Clearly the time element in this procedure would prohibit the calculation on any but the fastest computers. Turning to the FFT the operation would only take about 21 million operations or an improvement in excess of 5 orders of magnitude in computational time. The FFT thus brings the study of turbulence signals within the bounds of computational feasibility.

The ability to determine the power spectral density function which is computed from the Fourier-transform and which is such an important aspect of turbulence analysis is made possible through the use of the FFT algorithm.

III. NAVAL POSTGRADUATE SCHOOL DIGITIZATION FACILITY AND SPECTRAL ANALYSIS PROGRAMS

A. NPS COMPUTER FACILITIES FOR DIGITIZATION AND PSD ANALYSIS

The Naval Postgraduate School has two sophisticated computer systems which are used in the digitization and PSD analysis procedures. One is a Hybrid system which consists of an analog computer, COMCOR Ci 5000, which is electrically interfaced with a digital computer, XDS 9300. The XDS 9300 contains 34K bytes of core storage and uses an octal number base. This base consists of binary numbers made up of 3-bit digits. The second system is an IBM 360/67 which has a core storage of 762K bytes. It uses a hexidecimal number base which consists of binary numbers of 4-bit digits.

1. Hybrid Computer

The Hybrid computer facility is located on the fifth floor of Spangel Hall. The equipment disposition is shown in Figure 8. Though the individual user performs all computer operations, technical assistance is availabe from the Electrical Engineering Computer Laboratory staff. This facility is used for the analog-to-digital conversion of signals.

2. IBM 360/67 Computer

The large, digital computation facility is located on the ground floor of Ingersol Hall. Computer Center personnel handle all computer operations. Individual programs are input

Figure 8. Hybrid Computer Facility

to the computer through a card reader and requests for the subsequent input of programs, tapes, disk memory, etc. are submitted "over the counter" to computer personnel. A 24-hour operating schedule is usually maintained. All PSD analyses of digital tapes were performed on this computer.

B. HYBRID COMPUTER SYSTEM

Computer time on the Hybrid computer system is signed for in advance in the Electrical Engineering Computer Laboratory. The whole system is operated on a self-service basis; however, the computer center staff is available to assist during working hours. Computer time was easiest to obtain early in the term or between terms when the work-load was lightest. The facility is available 24 hours a day and as one becomes more proficient in equipment operations, night or week-end operations can be scheduled if week-day operations are precluded.

1. The Analog Computer (Ci 5000)

The Analog Computer contains the input points for raw signals input. It also has control features for signal amplification, selection of sampling rate and starting and stopping of the digitizing procedure. The two removable patch boards are the heart of the system. Individual patchboards have been reserved solely for analog-to-digital conversion (board #24).

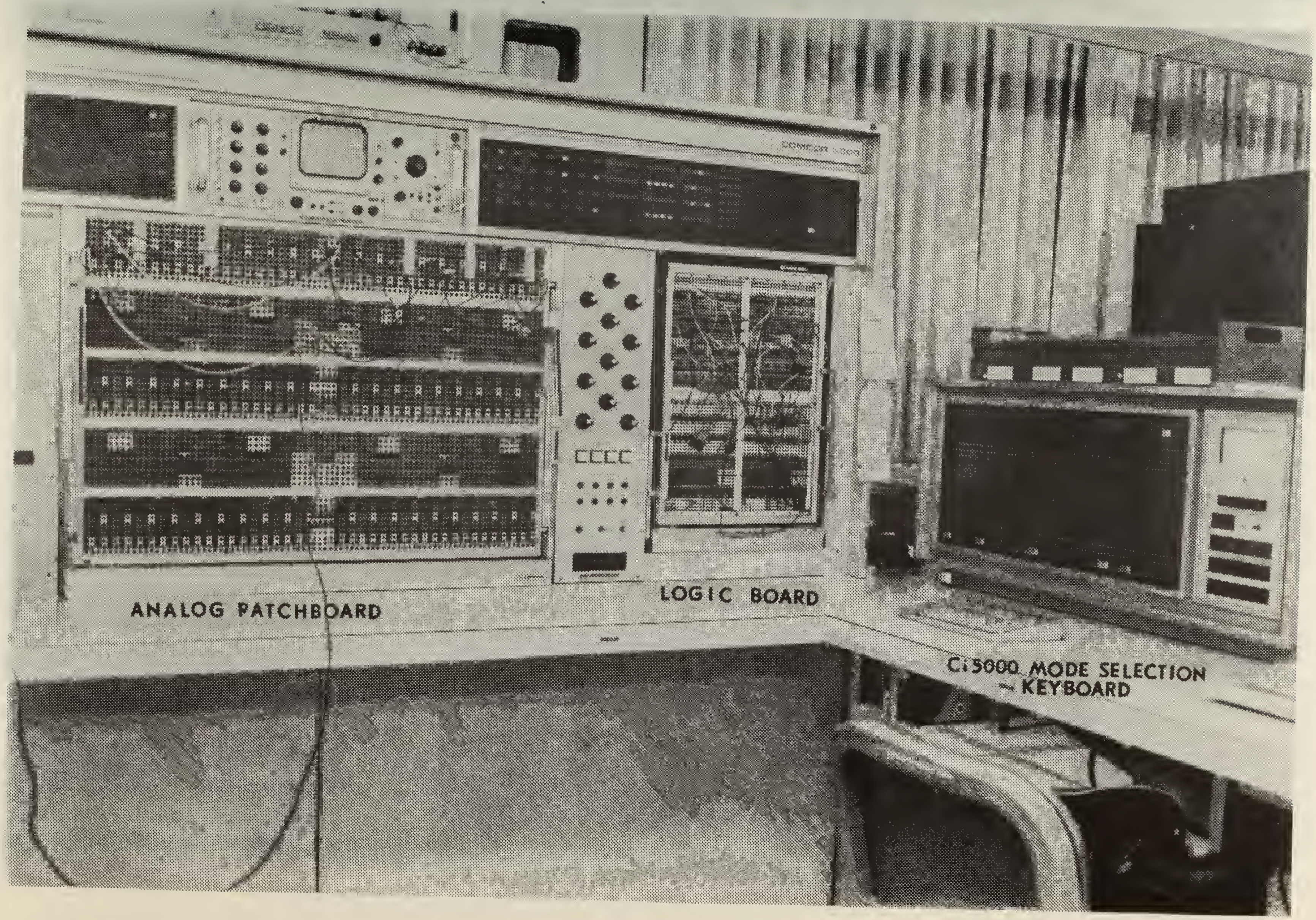


Figure. 9a. Ci 5000 Patchboards and Keyboard

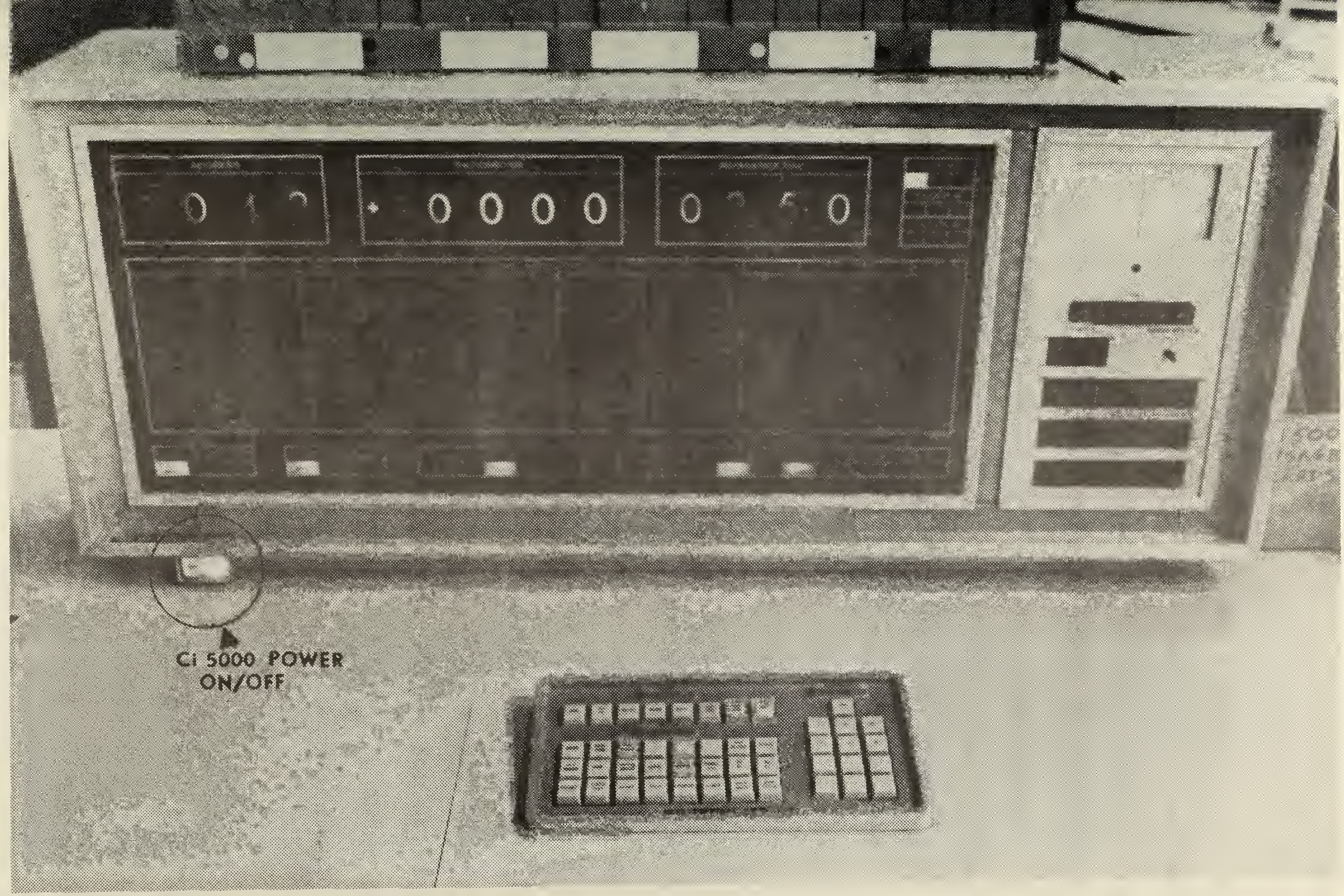


Figure 9b. Ci 5000 Keyboard and Power on Switch

The logic board, the smaller of the two patchboards, was utilized to set the sampling rate of the analog computer. The logic board with its patch is shown in Figures 9 and 17. The sampling rate was changed by turning a counter-like number display, below and to the left of the logic board, as shown in Figure 17. The counter operated a voltage divider which was electrically connected to a resistance input on the logic board. The resistance value at this point was changed by inserting a resister in one of 4 positions. The resistance value was then divided by the counter setting plus one to give the sampling rate (in samples per second) desired.

Example: $\frac{100\text{kc}}{(24+1)} = 4\text{Kc}$ or 4000 samples per second.

No external leads are connected to the logic board.

The analog patchboard is the input point for all external signals entering the analog computer. The input signal from a tape recorder or signal generator, after passing through signal conditioning equipment is input to one of the analog board amplifiers. (Figure 15 shows inputs going into amplifier A001). Various gain factors can be applied to the input signal to bring it up to a value optimum for utilization of the analog computers dynamic range. Figure 10 shows the keyboard of the analog computer which is used to control the desired operational mode of the analog computer.

2. Digital Computer (XSD 9300)

The heart of the digital computer system is the Xerox Data System 9300 Central Processor Unit (CPU). Two tape drive units, line printer, card reader and teletype unit are interfaced with the CPU. Figure 11 shows, diagrammatically, the

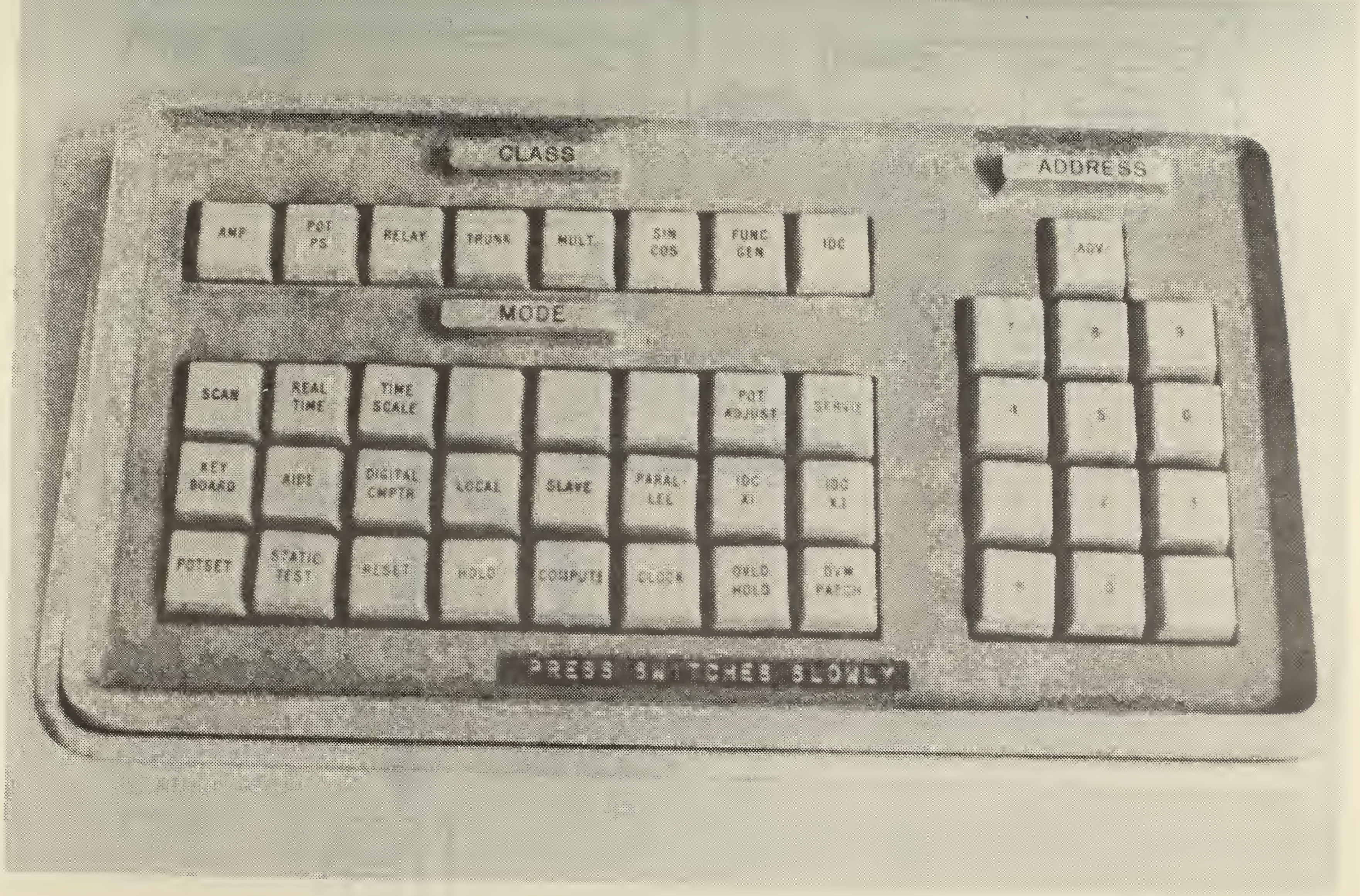


Figure 10. Ci 5000 Keyboard

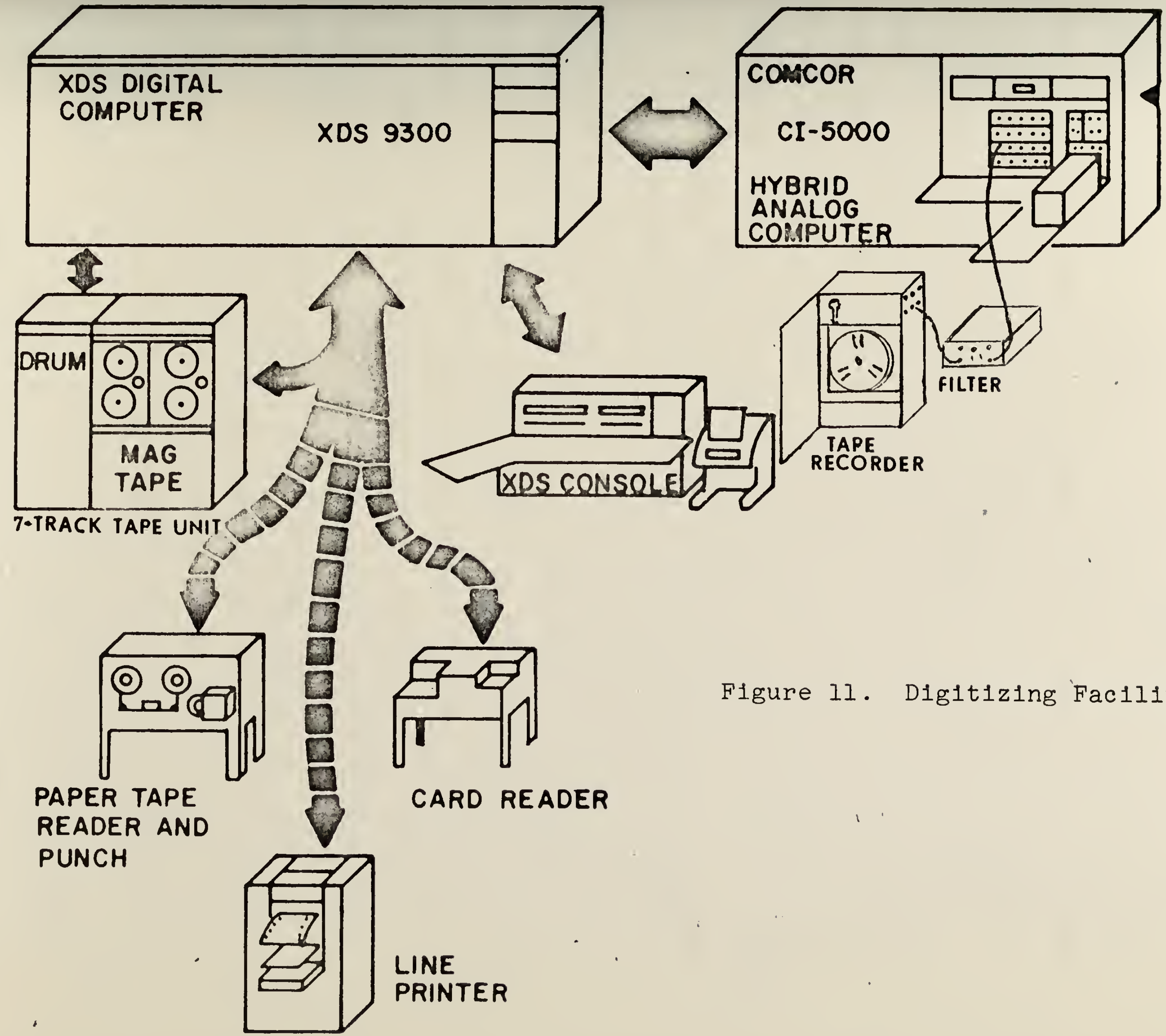


Figure 11. Digitizing Facility

equipment used in digitization. The tape drive units are used for data input from tape to computer or for data output from computer to tape; the line printer can be used for program listings and printing results; the card reader as program input for compiling programs; the teletype unit for input of parameters necessary to control the "multi-channel analog-to-digital Program." The CONTROL CONSOLE of the XDS 9300 is used to compile and to initialize the program. After the program has been compiled, the teletype is used to select one of seven digitizing options. The digital and analog computers are connected through the Analog-to-Digital Converter (ADC), (Fig.12).

3. Multi-Channel Analog-to-Digital Program

Several programs have been developed by the computer laboratory staff to control the XDS 9300 system during multi-channel digitizing operations. A card deck of the latest revision of the program is maintained in the computer lab. This program can be input from either cards or from a tape which has had the machine language program stored on it. The compiling sequence takes about five minutes using the card input and only about 30 seconds using the tape input.

Once the program has been compiled, one of the seven Program Control Options can be selected by typing one of the options followed by carriage RETURN (C/R) on the teletype (see Figure 13.). The Program Control Options are:

1. Enter new parameters. (NSAMP, NCHAN, NREC, ITAPE)

2. Start digitizing the analog input signals (Digitization actually starts when manual switch DSI on Ci 5000 is thrown to up position).

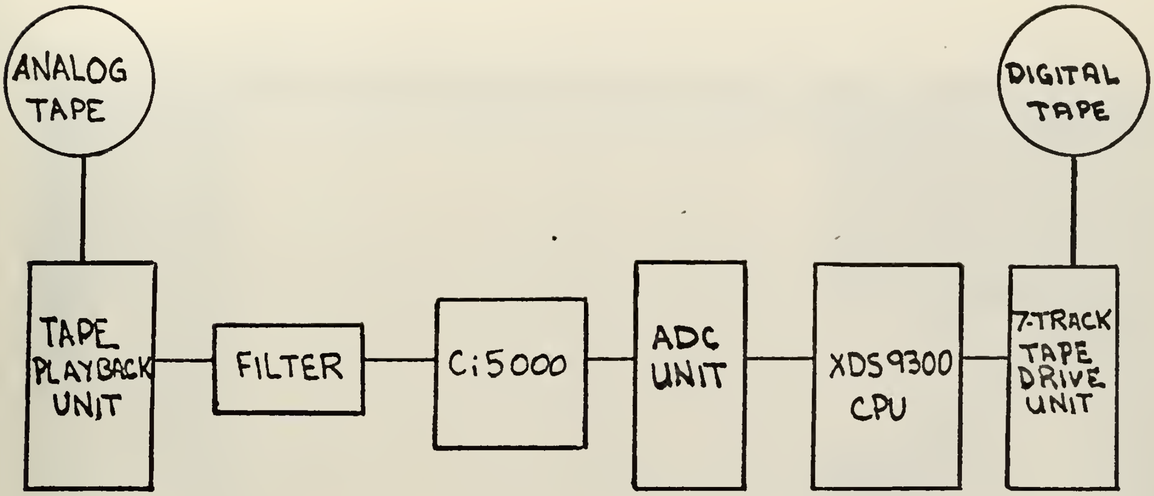


Figure 12. Block Diagram of Analog-to-Digital Conversion

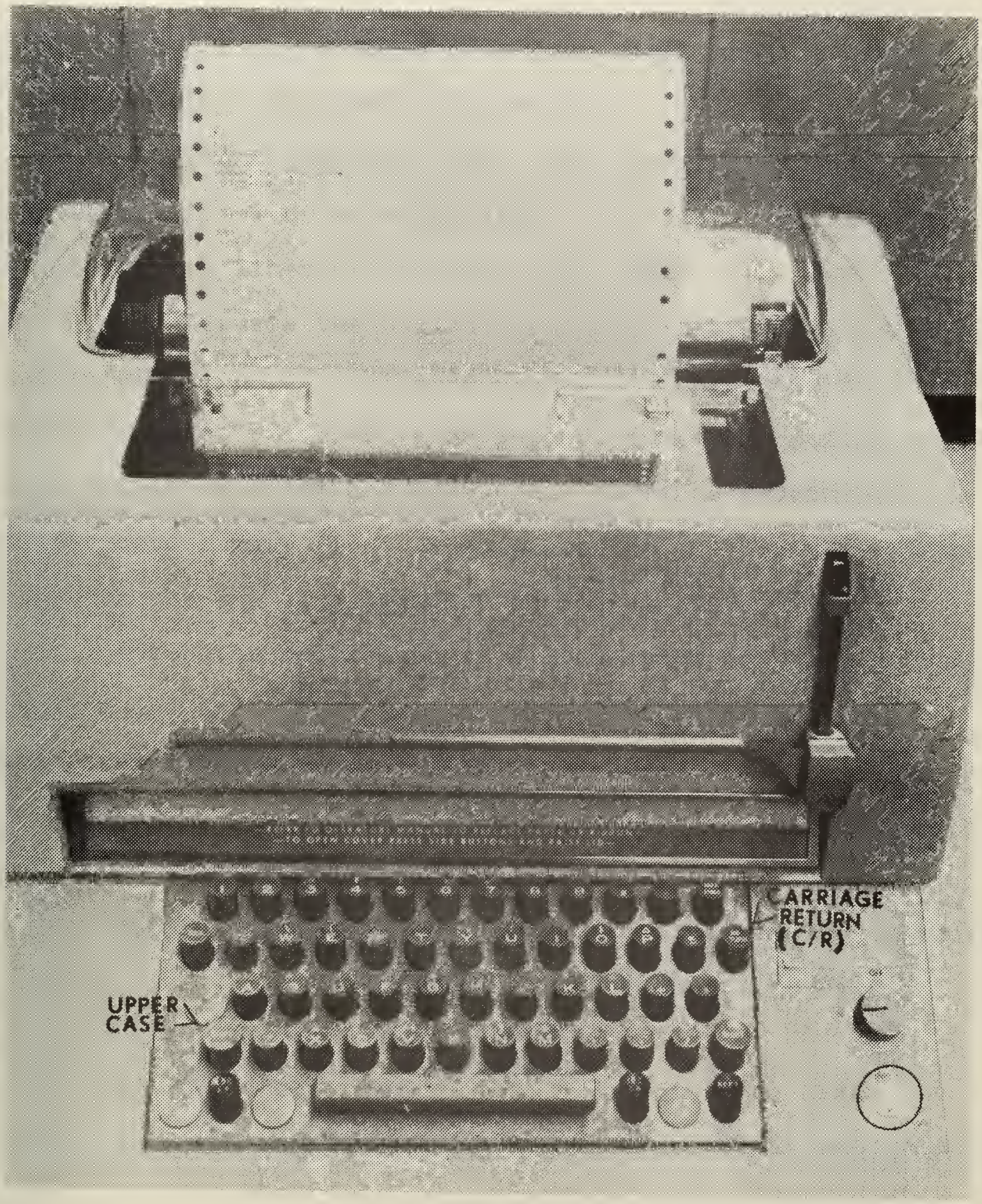


Figure 13. Teletype

3. Write End-of-File on tape.

4. Rewind tape to load point.

5. Skip files (the number of files to be skipped specified by teletype input of a four digit number, i.e. skip five files 0005 as shown in Figure 14).

6. Print digitized data (the number of <u>lines</u> to be printed specified by teletype input of a four digit number, i.e. print ten lines of digitized data 0010.

7. Actuate the Digital-to-Analog subroutine for the next single block of data encountered and input the data into the strip chart recorder (a subsequent computer comment types "start strip recorder and type C/R - carriage return"). The strip recorder must first be connected to the digital-to-analog output on the analog patchboard on the Ci 5000.

After the multi-channel digitization program has been compiled and the input light on the teletype is on, tape parameters input through the teletype are:

NREC - sets the maximum number of records to be digitized (see Figure 14).

NSAMP - sets the number of digitized samples contained in each record on the seven-track tape (see Figure 14).

ITAPE - sets the magnetic tape unit into which the digitized samples are sent. This number must be the same as the dial number selected on the front of the tape drive unit (see Figure 14).

NCHAN - sets the number of analog channels to be digitized (see Figure 14).

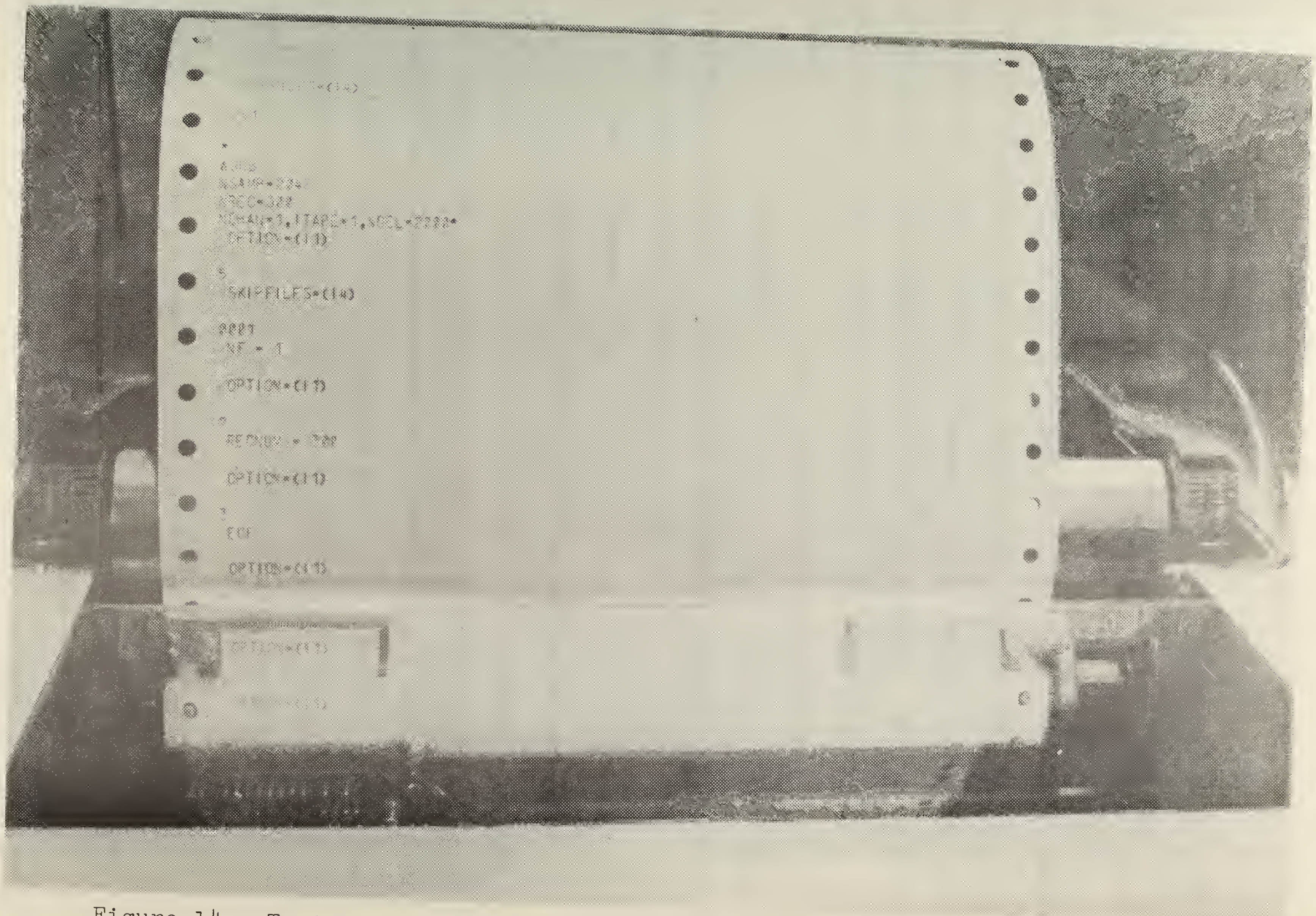


Figure 14. Typical Teletype Inputs for Analog-to-Digital Program Control

C. ANALOG- TO-DIGITAL OPERATIONS

1. Equipment Set Up

a. Analog Tape Recorder

The most commonly used record/playback device in the Oceanography Department is the Sangamo tape recorder Model 3562, which can record up to 14 tracks of data, and two edge-tracks of voice on one inch magnetic tape. Two modes of operation are possible: direct and FM. It is general practice to only use the FM mode, which records from D.C. to some upper limit depending on the tape speed. The FM mode is most noise free. Direct electronics are used for higher frequency data when no D.C. information is required.

The magnectic heads and tape rollers of the tape recorder should be cleaned with isopropyl alcohol after prolonged use to reduce noise during playback.

b. Filters

KHRON-HITE, Model 3321 filters have been most widely used in the past for filtering out unwanted high and low frequencies from a signal. They can either by connected singlely in a low-pass mode or in series as a band pass filter. After the filter mode has been selected the raw signal is input to the filter and the filter output connected to the analog patchboard as an input to the analog computer.

c. Analog Patchboard

The analog board should be patched as shown in Figure 15 if two channels are to be digitized. If only one channel is desired the signal input should be input into amplifier A001 only. Gain factors may be patched as desired.

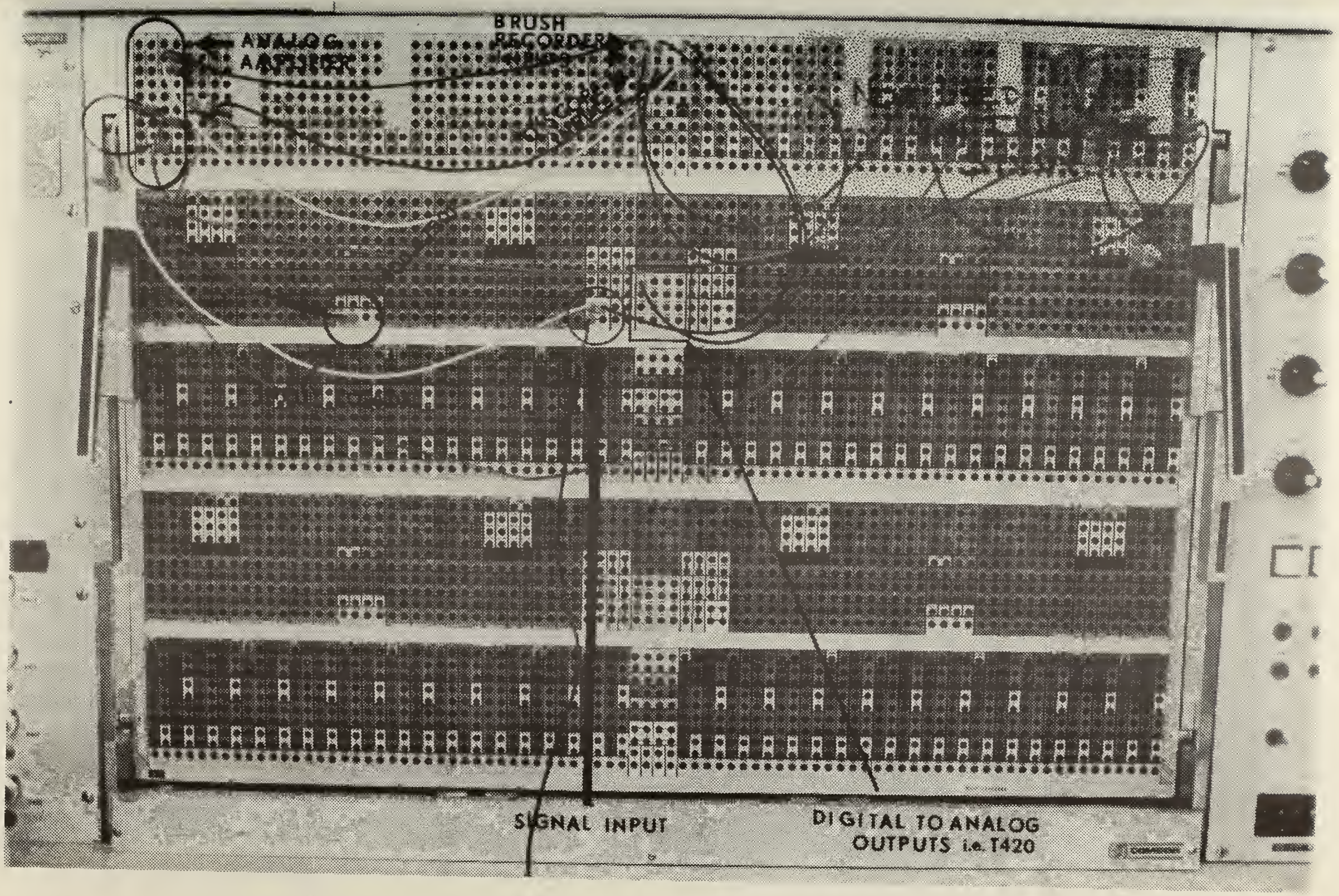


Figure 15. Analog Patchboard Used for the Analog-to-Digital Conversion of Electrical Signals

The outputs of each amplifier go to both the oscilloscope input points and the ADC input points (T500 and T501). If an analog playback of the digitized signal is desired, inputs from T420 and T421 (see Figure 15) must be made into the recorder inputs (1-8).

d. Logic Patchboard

The patch for the logic board is shown in Figure 17. This is for continuous mode digitizing in which digitization is started by throwing switch DS1. The only logic setting necessary is the selection of the required resistance value (Figure 17) which in conjuction with the wheel counter sets the sampling rate.

e. Oscilloscope

The "scale illumination" dial energizes the oscilloscope (see Figure 16). The sampling pulse was usually patched into "channel 1" by throwing switch Y1 to the left. This permitted continuous monitering of the sample pulse frequency and width. Dial DF00 (see Figure 17) should be set to Ms. 1 and the width of the sample pulse adjusted with this dial.

2. <u>Energizing the Ci 5000 Computer</u>

If the power on the CI 5000 has been secured (see power light in Figure 9b) it can be turned on by depressing the "on" switch. Next, the buttons KEYBOARD, POTSET and RESET are depressed in that order (see Figure 10).

3. <u>Energizing the XDS 9300</u>

The step by step procedure below should be followed in energizing the Hybrid computer system. The latest update to this procedure is kept on the XDS 9300 Control Console.

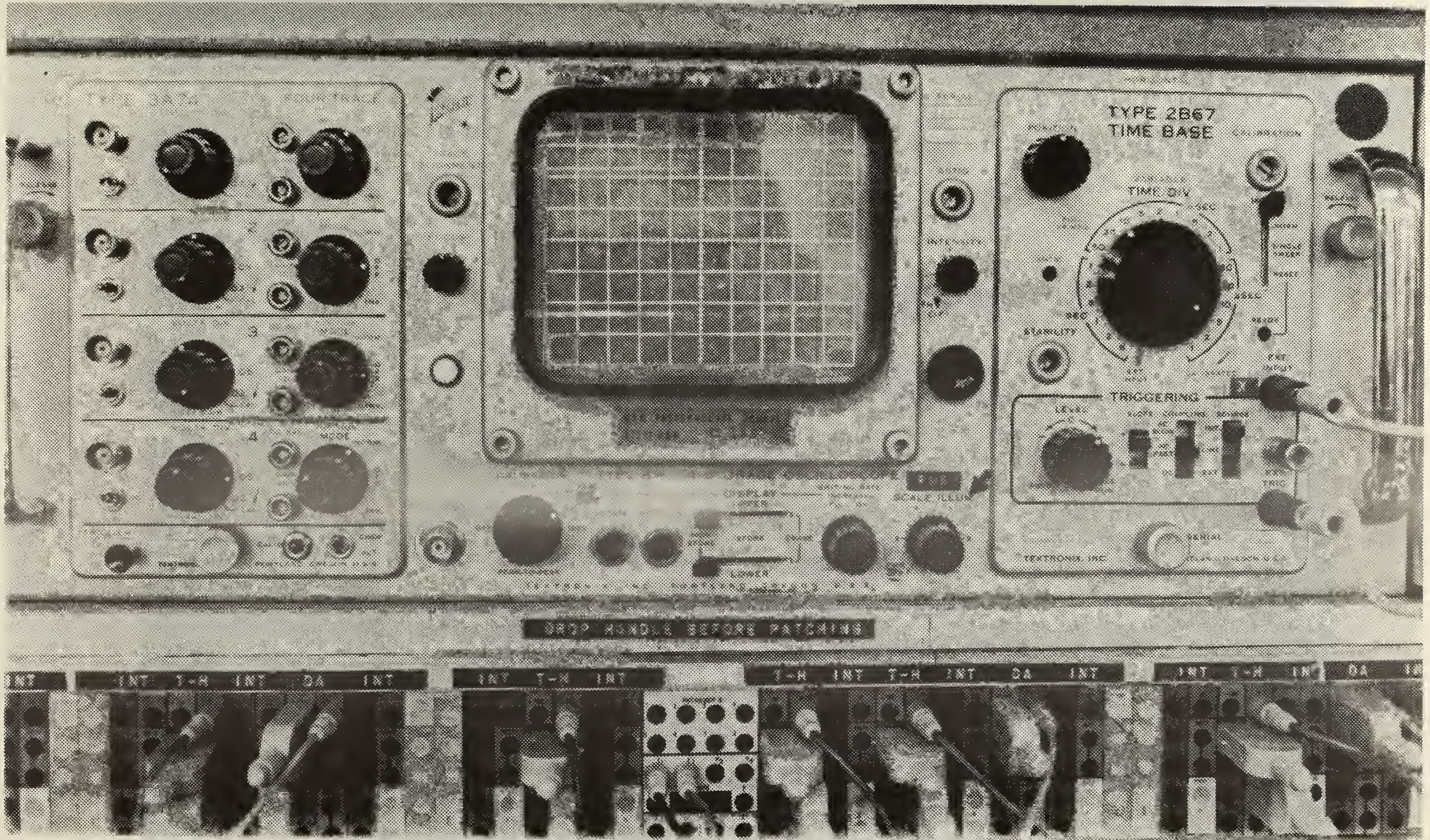


Figure 16. Ci 5000 Oscilloscope

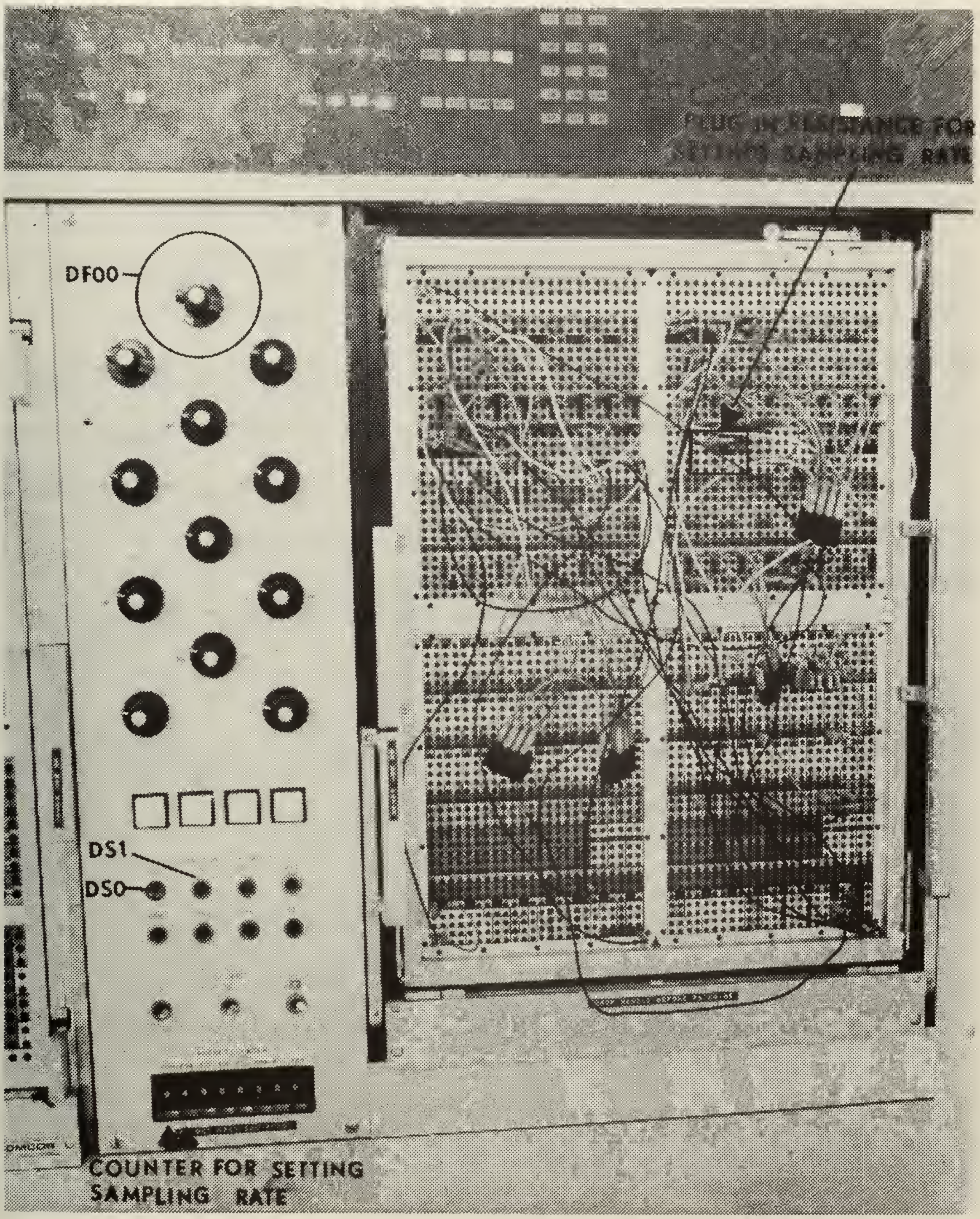


Figure 17. Ci 5000 Logic Board and Operating Switches

a. XDS 9300 Power on (if power is off). see Figure 18

1) Depress IDLE

2) With RESET depressed press POWER

3) Turn teletype switch to ON

4) Press CLEAR and CLEAR FLAGS simultaneously

b. Energizing the line Printer (if the READY light isn't on)

1) Insure that POWER button on line printer is on. If not, press POWER button.

2) When lower half of POWER is lit (within one min. after POWER pressed) press READY, NOTE: READY must be off to advance paper. A whole page is advanced with TOP of FORM and the paper is advanced a single line with SINGLE SPACE.

c. Card Reader (if A/D program being input from cards)

1) Place BOOT card in front of the JOB card in the A/D deck.

2) Put cards into card reader (see Figure 19) face down, top outward (toward you) so that column 1 is first into reader.

3) Put card reader press in order: POWER and START.

d.. Program Compile and Load

1) Ready the Line printer (sequence b above).

2) With cards in place ready the card reader (sequence c above).

3) At 9300 console turn off (by depressing) all SENSE switches which are lit.

4) Press in order: IDLE; RESET: RUN: CARDS.

5) Δ JOB is printed out by the teletype indicating compiling has begun.

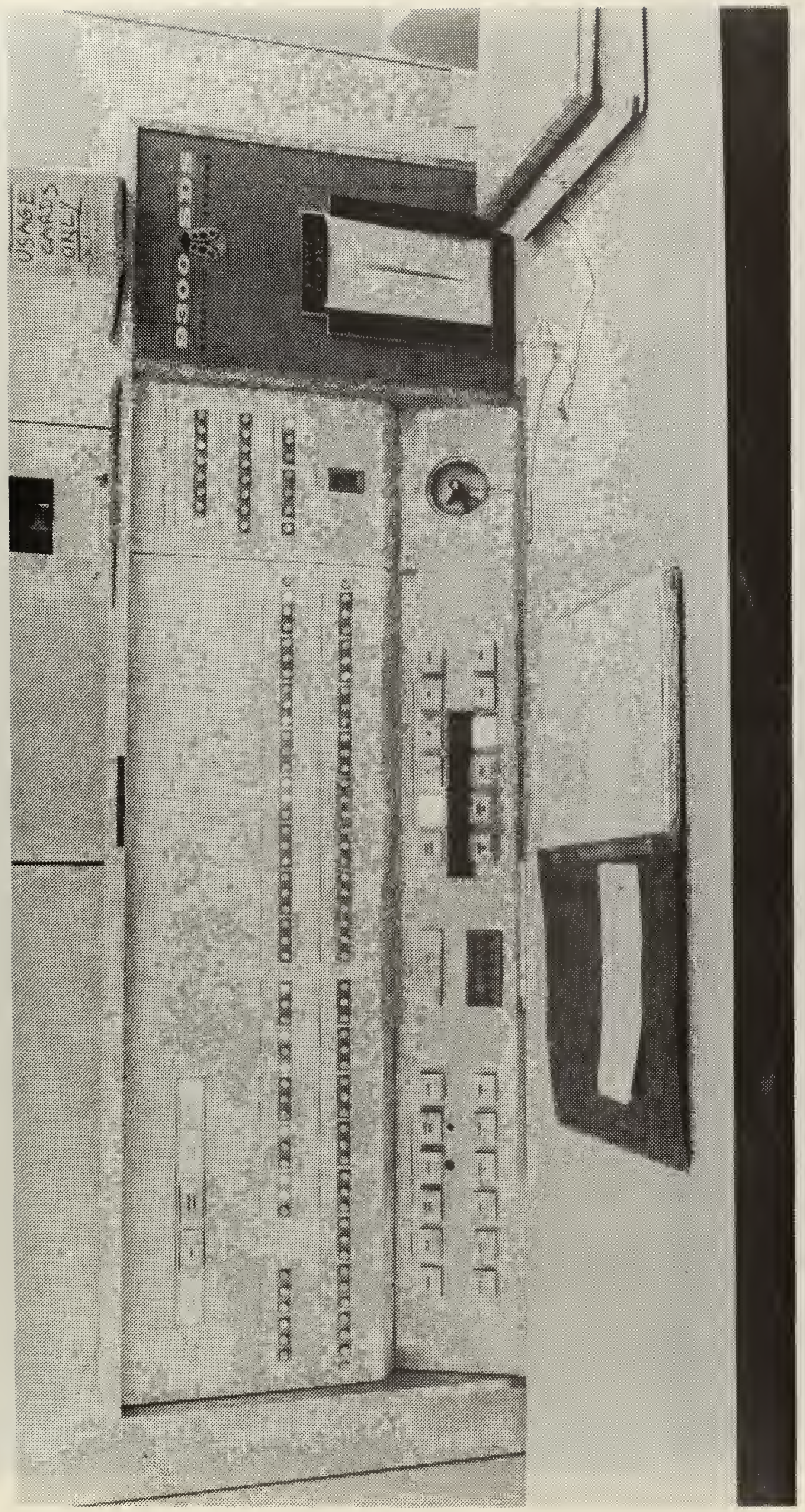


Figure 18. XDS 9300 Control Console

Figure 19. XDS 9300 Card Reader

6) Upon compilation, the teletype will type END OF ASSEMBLY. Shortly, thereafter, the INPUT light will go on, indicating the tape parameters can be typed in.

e. Equipment NOT-READY Condition

1) If a device is not ready when needed, it will be called out by a teletype message. Simply making the device ready will clear the pause in the operation.

2) In the event of a FEED-CHECK failure, inspect the card that caused the problem (the one on the bottom of the input hopper). If the card is found to be damaged on the leading edge, repunch the card and replace it. Put the unread cards back in the hopper. Ready the card reader by pressing RESET and START.

3) Line printer not-ready may be caused by its being out of paper.

f. XDS 9300 POWER OFF (after normal working-hours only)

1) Press IDLE.

2) Turn teletype switch to OFF.

3) Press IDLE.

4) With RESET depressed, press POWER.

4. <u>Mounting Magnetic Tapes</u>

It is very important to use currently certified tapes which are free of permanent errors. Scratches, tears, oil spots, etc. cause tape-writing errors. The only recourse is to write an END OF FILE on the tape at the end of the bad file and commence digitizing on the next section of tape.

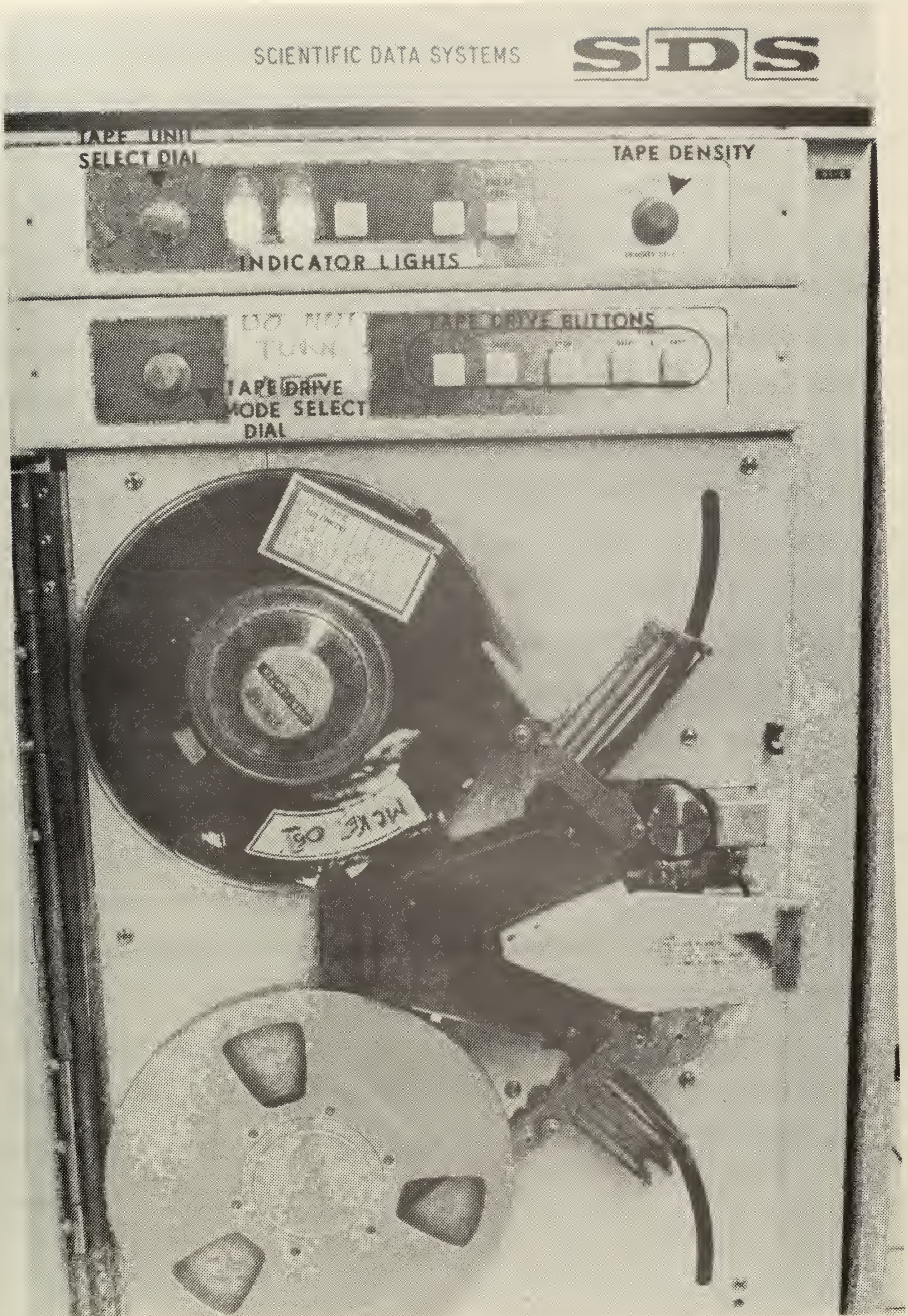


Figure 20. 7-Track Magnetic Tape Drive Unit

Figure 20 shows one of the two tape drive units which may be selected for tape mounting. Turn the mode dial to MANUAL READ. Before mounting the tape a circular plastic file WRITE RING must be inserted into the circular slot on the back of the seven-track tape being mounted. Mount the tape. If the ring is in place the FILE PROTECT, Figure 21, light on the top row of indicator lights will go out, indicating that data can be written on the tape. Tape threading instructions are posted on the inside of the protective doors enclosing the tape reels. Once the tape is secure around the take-up reel, it should be wound around at least six more times to prevent the tape from slipping off the take-up reel, when the main securing latch is closed. With the securing latch in place and the tape-idler-arm down, depress FORWARD-DRIVE. The tape will move forward and stop when it reaches the metallic strip called the LOAD POINT. The indicator light LOAD POINT on the top row of indicator lights will go on. After the tape has been mounted and advanced to the LOAD POINT, the tape density is selected by turning the DENSITY dial to either 256 or 556. This determines the recording density in bits per inch and is normally set to 556. The TAPE UNIT dial is turned to any one of several numbers, i.e. 1. (The teletype input parameter ITAPE must match this number, i.e. ITAPE = 1.) Finally, the mode dial is turned to automatic. The TAPE DRIVE UNIT is ready.

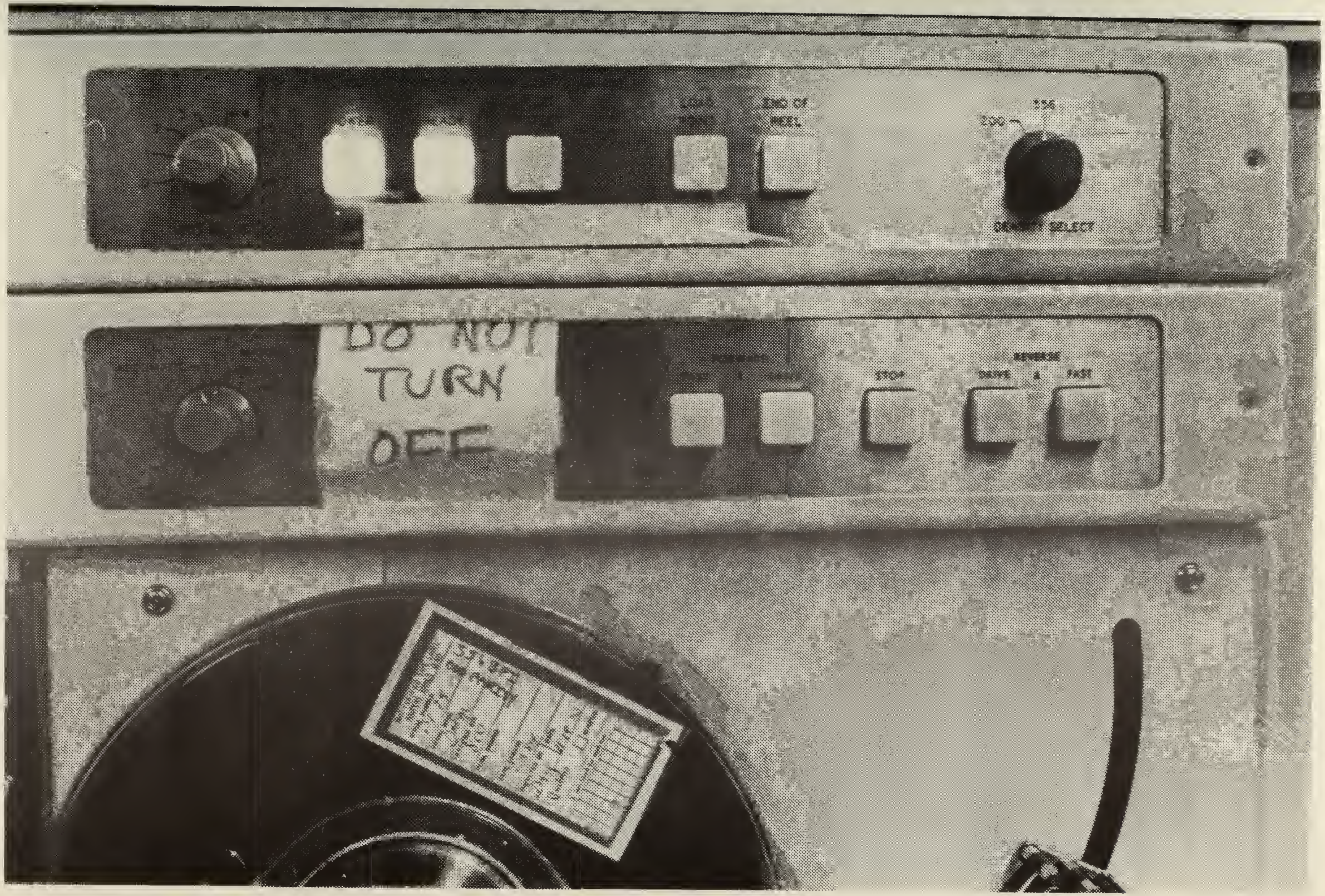


Figure 21 7-Track Tape Operating Dials and Buttons

5. Variable Tape Digitizing Parameters

Once the energizing and sampling sequences have been completed (the INPUT light on the teletype should be on), the tape options are entered through the teletype by typing a single digit number and then depressing the CARRIAGE RETURN, C/R button. In order to input new parameters, "1 C/R" would be typed.

Then the following example parameters could be typed in:

NSAMP = 2048 (C/R)

NREC = 100 (C/R)

NCHAN = 2, ITAPE = 1, NDEL = 2000 (C/R)

The computer would then type OPTION (I1), soliciting a further response from the user. If analog-to-digital collection was to commence, the programmer would type "2 (C/R)." If the Analog computer switch DS0 was in the UP position, digitization would begin when DSI was thrown to the UP position.

As would be expected, a high sampling rate would fill up 100 records faster than a lower sampling rate. Thus, if a sampling rate of 2000 SPS had been selected, and the above tape parameters had been selected, the digitization would result in 204,800 samples being collected and a total signal length of 102.4 seconds.

$$2048\ \frac{\text{SAMPLES}}{\text{RECORD}} \times 100\ \text{RECORDS} = 204{,}800\ \text{SAMPLES}$$

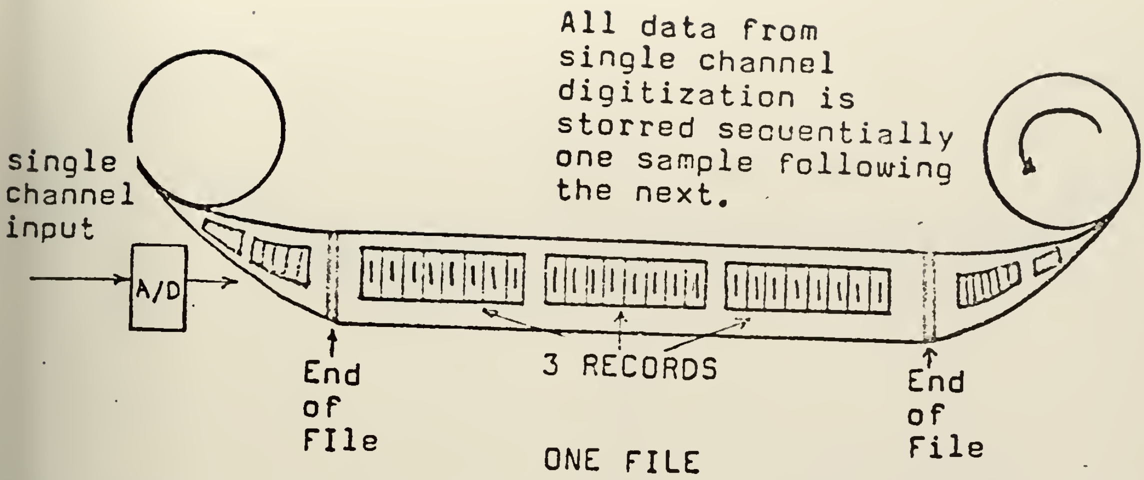
$$\frac{204{,}800\ \text{SAMPLES}}{2000\ \text{SAMPLES/SEC.} \times 2\ \text{CHAN}} = 51.2\ \text{SECONDS}$$

There would only be 102,400 samples of each channel; however, there were two channels being digitized simultaneously. Figure 22 shows how digitized data for one and two channel digitization would be formatted on a seven-track tape.

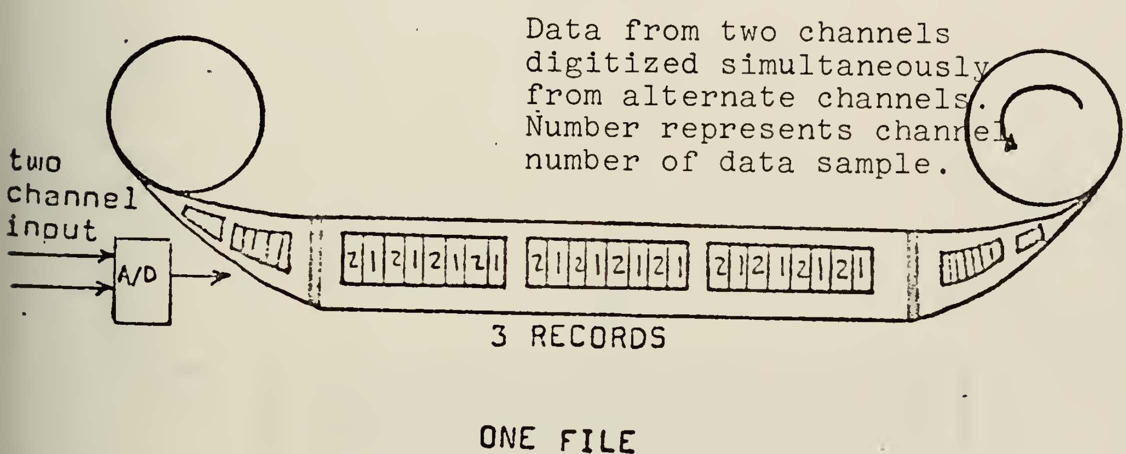
D. CONVERT PROGRAM

The CONVERT PROGRAM converts the seven-track octal base data samples into nine-track hexidecimal samples. The basic character of the data samples being in blocks on the magnetic tape is maintained; however, two additional parameters (see Figure 23) are affixed to the beginning of each block by the CONVERT program. These are the numerical values KMAX which is set equal to the maximum number of samples per block and NCHAN, the number of channels of analog data digitized on the seven-track tape. This change in the block format, the change in number base and the addition of two new parameters is necessary to put the tape in the proper format for input into the FTOR program.

Figure 24 shows the CONVERT procedure in flow-chart form. The program processes all of the records in a file and the number of files to be converted is specified in the IF statement which compares the number of files completed with the number specified for processing. The number in parenthesis is set to the number of files to be converted. The JOB CONTROL LANGUAGE (JCL) cards which follow the CONVERT PROGRAM specify which files on a multi-file tape are to be processed. Jones [Ref. 5 Appendix D] gives the typical JCL cards used to convert five files.



a) Single Channel Tape Formatting - 3 Records with 8 samples per Record ($N=2^3$)



b) Dual Channel Tape Formatting - Records with 8 samples per Record

Figure 22. 7-Track Tape Formatting for Single and Dual Channel Digitization

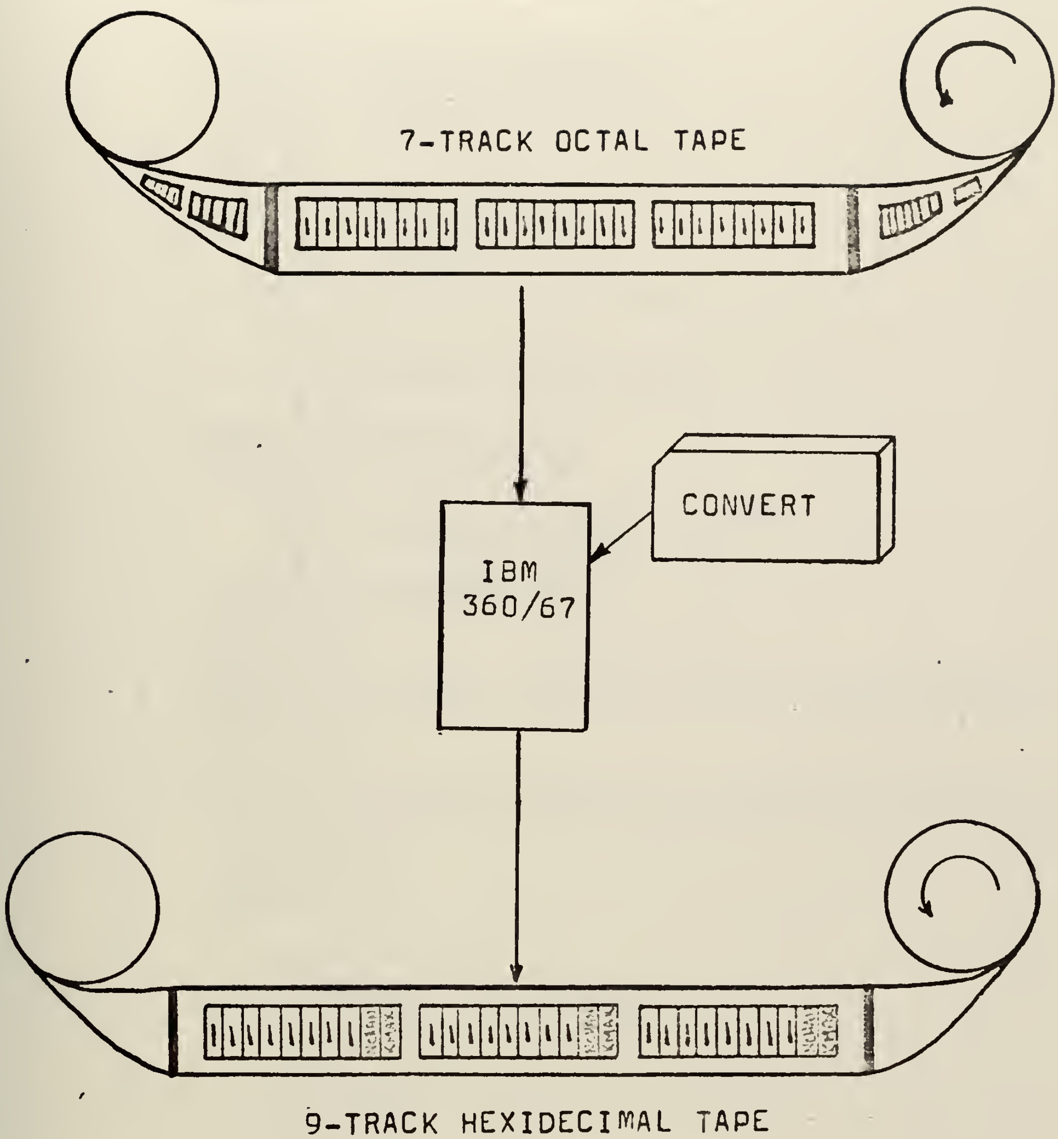


Figure 23. 9-Track Tape Formatting for FTOR Program

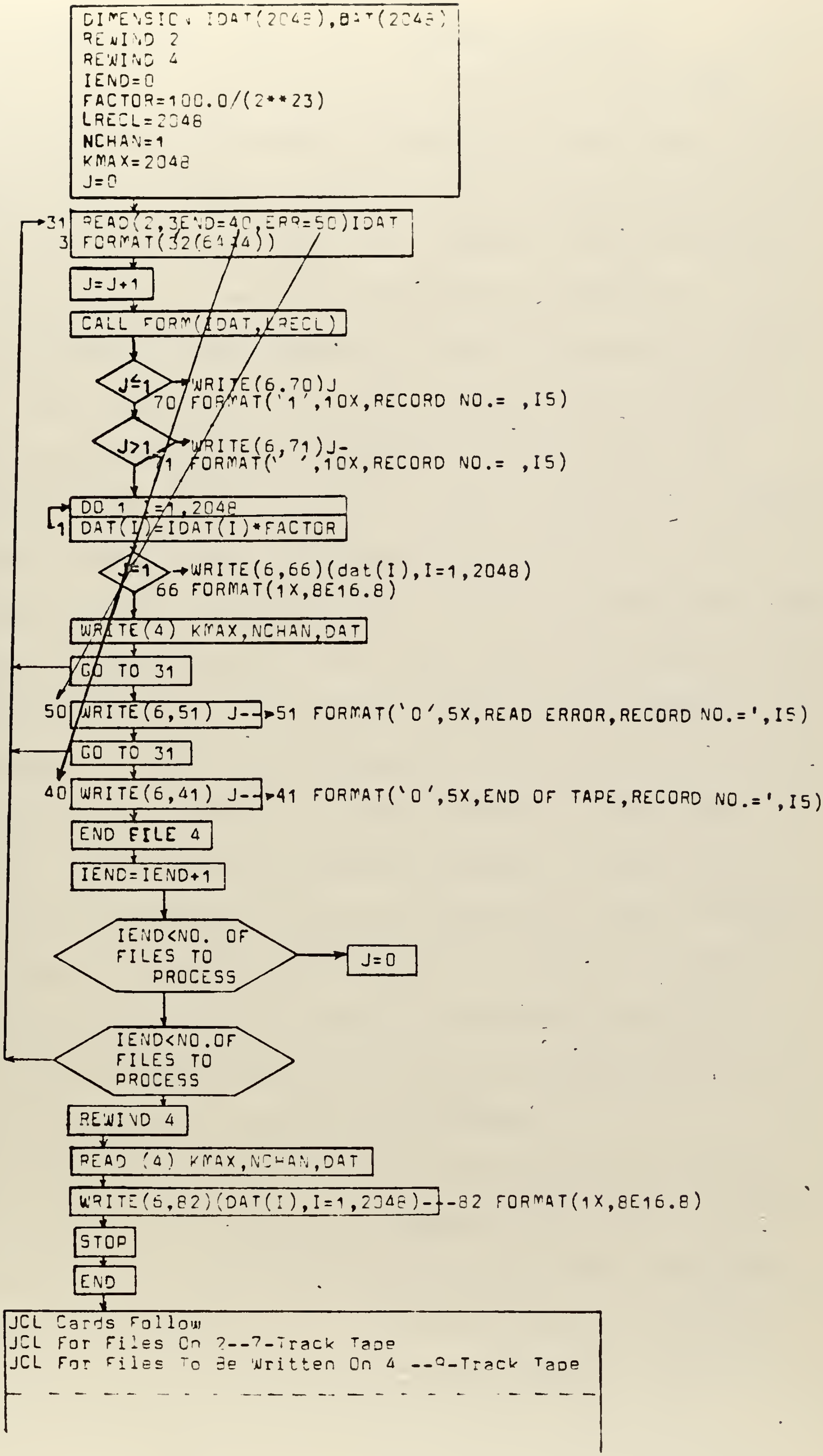


Figure 24. CONVERT Flow Chart

E. NAVAL POSTGRADUATE SCHOOL PSD COMPUTER PROGRAMS

Several computer subroutines are available on the IBM 360/67 for computing the Fourier transform of digitized input signals. Most programs, however, are designed for the input of relatively small data sets, usually from Hollerith cards. These programs could be designed to compute the Fourier transform of large amounts of data on magnetic tapes; however, a very powerful series of programs is available in the Naval Postgraduate School computer library which will compute the FFT of large data sets. These programs are UBCFTOR UBCSCOR and UBCFCPL which are stored under files one, two and three respectively on tape NPS 216. The only particular format requirement for input data is that the digitized samples must be arranged in groups of an integral power of two samples in each "record." A record is 2^j samples where j is an integer.. Many records may be found on a single magnetic tape (see Figure 22). Since this FFT program package is especially well suited for analyzing large digitized data sets stored on magnetic tape, it serves as an essential component of the Naval Postgraduate School's signal processing capability. Figure 25 shows schematically the complete digitization and PSD analysis procedure using these programs.

The following description of the FFT package was taken from Dobson [Ref. 6]. Editorial changes were made to update the information with the Naval Postgraduate School facility. The programs were written by J. F. Garett and J. R. Wilson while students at the Institute of Oceanography at the

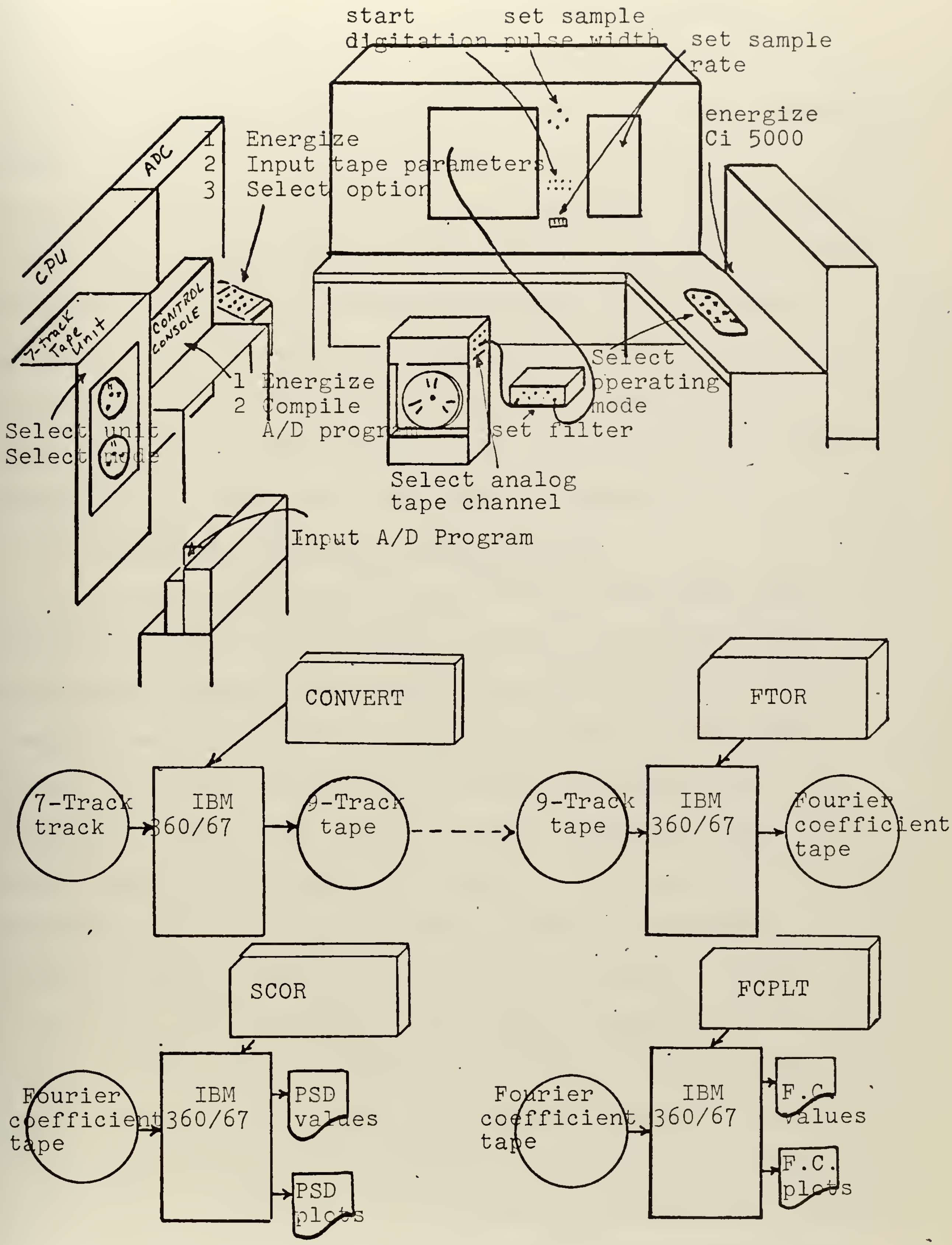


Figure 25. Schematic Diagram of Complete Digitization and PSD Analysis Procedure

University of British Columbia. The UBCFTOR program, written in Fortran IV, instructs the IBM 360 to read the digital data tapes, store a block of data into the computer memory, compute the FFT of the data, and then store the resulting Fourier coefficients on another tape. From these coefficients the UBCSCOR program computes the spectral density for single channel, or the cross-spectral density between selected pairs of channels. The spectral density values may be "output" in either tabular form or in graphic form on the Calcomp plotter. The UBCFCPLT plots the amplitudes of the Fourier coefficients as a function of frequency.

1. Fourier Coefficient Program UBCFTOR

This program calls several subroutines which read in data from the digital tapes generated in the A/D phase. The maximum number of data points which can be stored in the computer memory under this program are 8192 or 2^{13} samples. This restriction was imposed by PKFORT, the subroutine which computes the FFT. The details of this subroutine are listed under PKFORT in the computer library. For optimal efficiency PKFORT has been designed to read in data blocks which contain an integral power of two (i.e. 2^j) samples. PKFORT is then called and the Fourier transform is computed, which results in 2^{j-1} Fourier coefficients. Since only one block of data is analyzed at a time, the original signal is divided into short, sequential sections of signal and a new length of record T transpires. The resulting coefficients are stored on another magnetic tape and they become the basis for the

spectra. For this block of data, the highest frequency for which a coefficient has been computed is the Nyquist frequency fs/2, where fs is the sampling frequency. The lowest frequency which can be resolved for this data block is 1/T or 2^j/fs.

The Fourier coefficients are written into blocks on a "coefficient" tape which also contains identifying information such as the block number, sampling frequency, number of samples in each block, etc. Upon completion of the transformation and writing sequence for one block, the next block of data is read from the digitized tape. The sequence is repeated for as many blocks as specified on the input data cards, until an end-of-file mark is sensed on the digitized tape or until a blank card is encountered in the input data cards.

2. Spectral Analysis Program - SCOR

Once the Fourier coefficients have been computed, the next step is to convert these values into spectral values. The program SCOR reads the coefficient tape generated by FTOR and averages the PSD values over 32 bandwidths. The data card section of the SCOR program specifies the number of channels, the number of blocks to be analyzed within a file, the block number where the analysis is to begin, whether the spectra for a single channel or the cross-spectra between channels is to be computed, the type of bandwidth desired (constant or logarithmic), etc. Just as with FTOR procedures, SCOR reads and analyzes only one block of coefficients at a time.

If smoothing of the coefficients is desired, various smoothing functions may be selected. The "hanning" option performs a three point running average on the data with weights of -1/4, 1/2, -1/4. References 2 and 6, contain further details of the smoothing functions.

If individual Fourier coefficients of two channels are $R_1 + iI$, and $R_2 + iI_2$, where $i = (-1)^{1/2}$ and τ equals the block length in seconds (and $\delta F = 1/\tau$ where δf is the bandwidth between Fourier coefficients) then the power spectral density is given by:

$$\phi_{11}(f) = \frac{\tau(R_1^2 + I_1^2)}{2} = \frac{R_1^2 + I_1^2}{2\delta f}$$

and
$$\phi_{22}(f) = \frac{\tau(R_2^2 + I_2^2)}{2} = \frac{R_2^2 + I_2^2}{2\delta f}$$

The cross-spectral values are given by the co-spectrum:

$$C_0(f) = \frac{\tau(R_1R_2 + I_1I_2)}{2} = \frac{R_1R_2 + I_1I_2}{2\delta f}$$

and the quad-spectrum:

$$Qu_{12}(f) = \frac{\tau(R_2R_1 - I_2I_1)}{2} = \frac{R_2R_1 - R_2R_1}{2\delta f}$$

The factor of 2 in the above equations makes the integral under the power spectrum over positive frequencies equal to the signal variance. Phase corrections can be made at this point to correct for instrument phase shifting and then recalculating the cross-spectra.

After all the required power and cross-spectra for a block have been computed and stored, the sequence is repeated until the number of blocks requested have been analyzed or until an end-of-file mark is sensed on the coefficient tape.

Then the program averages the spectral estimates both over the number of blocks processed and over the analysis bandwidth requested in the input data cards. The standard deviation from each mean is computed in a similar procedure from the formula

$$\sigma = \left\{ \frac{\sum_{N} (R_n - \bar{R})}{N-1} \right\}^{1/2}$$

where N is the number of samples used. Other useful information is computed and printed in either tabular form or graphic form as specified by the investigator.

F. IBM 360/67 TAPE OPERATIONS

1. Job Control Language

Job Control Language (JCL) cards provide the computer with information on tape mount number, tape identification, disposition of tape; identity of tape data file to process and order of data on the tape. A typical JCL card used for tape processing would be:

```
//GO.FT04F001 DD UNIT=2400,VOL=SER=NPS185,LABEL=(1,SL),
   DSNAME=MCKE01,DISP=(NEW,KEEP), DCB = (DEN=2
   RECFM=VS, BLKSIZE = 8204)
```

GO - This group indicates data is going to be classified under the subsequent identify parameters.

FT04 - This group indicates the tape is to be placed on logical unit (in this case a tape mount) four. Also the two digit number (04) must match the number specified in the READ or WRITE instruction in the program (i.e. READ (4) KMAX, NCHAN, DATA). The JCL does not determine whether "reading" or "writing" is to take place. The program statement which indicates the logical unit (tape mount) also indicates the process (READ or WRITE) desired.

F001 - This group indicates the sequential number of passes through the tape which has occurred up to this point. In this case it is the first pass.

UNIT = 2400 - This group indicates the particular unit will be a nine-track tape. The designation for a seven-track tape is UNIT = 2400 -1.

VOL = SER = NPS 185 - This group indicates that the tape has the external computer center marking NPS185. If a seven-track tape were used it could have a user specified name (i.e. MCKE01).

LABEL = MCKE01 - This group specified the Data Set Name (DSNAMÉ) of the data specified by the LABEL group. This name may be changed for each different file of data; however, such a procedure is quite time consuming. It is much quicker to use the same DSNAME for a whole tape. It is important that when ever a data set is created under a particular name, it must always be referred to by the same name when reading from the tape.

DISP = (NEW, KEEP) - This group specified the disposition of the tape when processing (either "reading" or "writing"). This implies it is a newly created tape (NEW) and it is to be saved (KEEP). If an old tape were to be read and the data saved, the selection would be DISP = (OLD, KEEP).

DCB = (DEN = 2, RECFM = VS, BLKSIZE = 8204) This group is the Data Control Block. The most often used density (DEN) is 556 bytes per inch which is denoted by 2. This corresponds with the manual setting of the DENSITY dial on the face of the seven-track tape drive unit. The tape record format (RECFM) is variable spaced (VS) when recording and digitizing on the seven-track tape. The block size (BLKSIZE) specifies the exact number of bytes in one block of data. If 2048 words per block were considered, four bytes per word would result in 8192 bytes per block. If, as in the CONVERT program, two words from KMAX and one word from NCHAN give an additional 12 bytes, the total is 8204 bytes per block.

2. Multi-file Tape Operations

Most frequently, a digital tape containing several files of data was generated. Under normal IBM 360 procedures, as with the CONVERT PROGRAM, the JCL cards are used to indicate which files of data are to be processed. But, in the FTOR and SCOR programs, JCL cards had to be included for every file, even though the file was not processed. This variation to the usual JCL procedures was necessary because of the SKIP-FILE subroutine in both programs which caused the files that were not specified for analysis to be skipped.

An example of the JCL cards for the normal processing of three files of data; where it was desired to skip the second file on a tape, was given by Jones [Ref. 5, pg. 46].

3. Multi-Volume Tape Operations

If one long file of data extends onto another tape and processing of this long data file is desired, multi-volume tape procedure becomes quite complicated when using multi-tape programs, its use should be avoided. The average seven- and nine-track tapes used can hold more than 1400 records of 2048 samples per record. This gives a capacity for more than two and a half million samples per tape.

G. PREPARATION OF CARDS AND TAPES FOR PSD ANALYSIS

1. JCL Cards For FTOR

This resume of JCL techniques for using the FTOR program are taken from Jones (1971). The following procedure assumes also that NPS 216 will be used with its KMAX set to 256. If changes are made to KMAX and the program is taken from another tape (as discussed under Modifications to FTOR), the same argument would be valid.

The FTOR program is stored on NPS 216 in File 1. The program is also storred on disk (see Wilson, et.al., (1969)). To use FTOR directly from the tape, the following cards should be used.

```
----JOB CARD ------------------------------------------
//FORT. SYSIN DD UNIT = 2400, VOL = SER = NPS 216, DISP = OLD,
//       LABEL = (1,SL), DSN = UBCFTOR
```

---JCL CARDS FOR INPUT & OUTPUT TAPES -----------------------

//GO. SYSIN DD*

---CONTROL CARDS FOR INPUT & OUTPUT PARAMETERS --------------

(BLANK CARD)

/*(See Wilson, et. al. [Ref. 7, Pg 33a and 33b]).

If the FTOR deck is used, the following cards should be used:

---JOB --

// FORT. SYSIN DD*

-----FTOR PROGRAM DECK -------------------------------------

-----JCL CARDS FOR INPUT & OUTPUT TAPES ---------------------

//GO. SYSIN DD*

---CONTROL CARDS FOR INPUT PARAMETERS -----------------------

(BLANK CARD)

/*

Jones [Ref. 5 p. 70] gives a good example of how the input cards for a complete FTOR run should appear. His format assumes that the FTOR program is being input from a deck of cards.

The calibration card referred to has a place for a primary and secondary channel. When the complete signal comes from only one "primary" input, the primary channel calibration card is the only one used. If an instrument, such as the sonic anemometer is used, signals need to be added vectorially to give the actual signal. The FTOR program allows for the input of two separate digital signals. The calibration cards permits these signals to be added vectorially and the spectrum computed for a single signal. The alphameric

section of this card allows for signal, parameter identification, such as sampling rate, filter setting, analog tape channel number, etc. to be storred with the Fourier coefficients. This alphameric section is printed as the title on all output graphs.

A problem might arise from the use of this alphameric section when this title is applied to long data lengths. When short sections of the signal are analyzed with SCOR, this title information is printed on all graphs. Thus, a slight bookkeeping problem occurs when thirty graphs are printed with the same title. Fortunately, the computer center delivers graphs in a continuous roll. Identification is made by finding the starting point and then numbering them sequentially from that point.

As mentioned previously, the SKIPFILE subroutine in FTOR made it necessary to include JCL cards for every file on the nine-track tape. This is a variation to the normal IBM 360 procedure in which the JCL card specifies which file to process. The SKIPFILE routine in FTOR, SCOR and FCPLOT allows for the internal selection of data files, based on the parameters on the data cards which follow the tape JCL cards in the input deck.

2. <u>Modifications of FTOR Program</u>

Jones (1971) found it was necessary to make several changes so that FTOR would be compatible with the block size specified by KMAX. It was discovered that the FTOR program on NPS 216 was written for a block size of 256 words. The

change involved setting KMAX, Card 18, equal to the number of words in one block of data. Also the KMAX dimension statement, card 23, was similarly changed.

This change was made by inserting the new card into the program deck. Then, in order that the whole FTOR deck need not be input each time, the modified program was written into File 1 of tape NPS 223. In order to reduce the number of different tapes used, the SCOR program was written into File 2 of NPS 223. If too many different tapes are used, confusion arises in mounting tapes. Thus, the modified FTOR and SCOR could be called from the same tape by the use of several cards rather than manipulating an entire deck of cards.

3. JCL For SCOR

The cards required for the "running" of the SCOR program follow the same order as those required for FTOR. The major changes are:

```
LABEL = (2,SL)  The program is stored in File 2
DSN = UBCSCOR   The data set name is UBCSCOR
JCL Cards for tapes must be changed to read the
FTOR coefficients from the FTOR generated tape.
Control cards for analysis must be appropriate for
SCOR program. (see Wilson, et.al. (1969) for
details).
```

Jones, [Ref. 5 p. 71], gives a good example of how the input cards for a complete SCOR run should appear. His format assumes the SCOR program is input from NPS216. This tape can be used without the KMAX changes needed for FTOR, because the input to the SCOR program was the Fourier coefficient tape which had a uniform format, regardless of the block size used in the nine-track digital input tape.

The selection of a uniform set of axes should be anticipated so that graphic outputs can be overlaid for easy comparison of spectra. If logarithmic bandwidths (exponential) are selected, a constant bandwidth bar appears on the log-log plot due to the exponential nature of the axes.

Jones [Ref. 5], points out that the SUBSEQUENT ICMAX CARDS referred to by Wilson, et. al. [Ref. 7, pg. 36] indicate what type of graphic output is desired: spectra, cross-spectra or both. The spectrum of a single channel is specified as if it is a cross-specturm of that channel with itself. For two channels (1 and 2), a spectrum of channel one and a spectrum of channel two are obtained by the following entries on these data cards:

First card:	1	1	2
Second card:	2	2	

The cross-spectrum is computed only is each channel appears in the list on the other channels card as seen with the first card above.

4. JCL for FCPLT

This program is used to get a Calcomp plot of the amplitude of the Fourier coefficients as a function of the frequency. The cards required for "running" the program follow the same order as required by FTOR and SCOR. The major changes required are:

a. LABEL = (3,SL) This program is stored in File 3.

b. DSN = UBCFCPLT The data set name is UBCFCPLT.

c. JCL cards for tapes may be the same as those used for SCOR if the same data files are to be used.

d. Control cards for analysis parameters required must be appropriate for FCPLT as specified in Wilson, et.al. [Ref. 7, pg. 59].

Jones [Ref. 5, pg. 72] gives an excellent example of how the input cards for a complete SCOR run should appear. Figure 26 was prepared to further clarify the JCL cards needed for each program.

5. Suggestions for Efficient Tape Processing

Since the techniques involved in running this program can be quite time-consuming, much attention was devoted to the problem of optimizing the number of runs per day. Another problem was the low priority for these programs. (Priorities range from the highest priority, class A to the lowest, class K). The low priority resulted because of two factors. First, the program computation time (specified on the green JOB card by TIME = MM,SS, where M is minutes and S is seconds), in most cases, took more than four minutes of Control Processor Unit (CPU) time. The second factor was that the program required using tape input and output units which are automatically put into a lower class because of the time required to mount them and access the information on them. If the programs and the large quantity of digitized signals could be completely stored on magnetic disks, higher priorities could be achieved. This procedure would save time if repeated analyses were to be conducted on a data set which did not change. In this study many different signals were studied, which implied the data set varied from one signal to another.

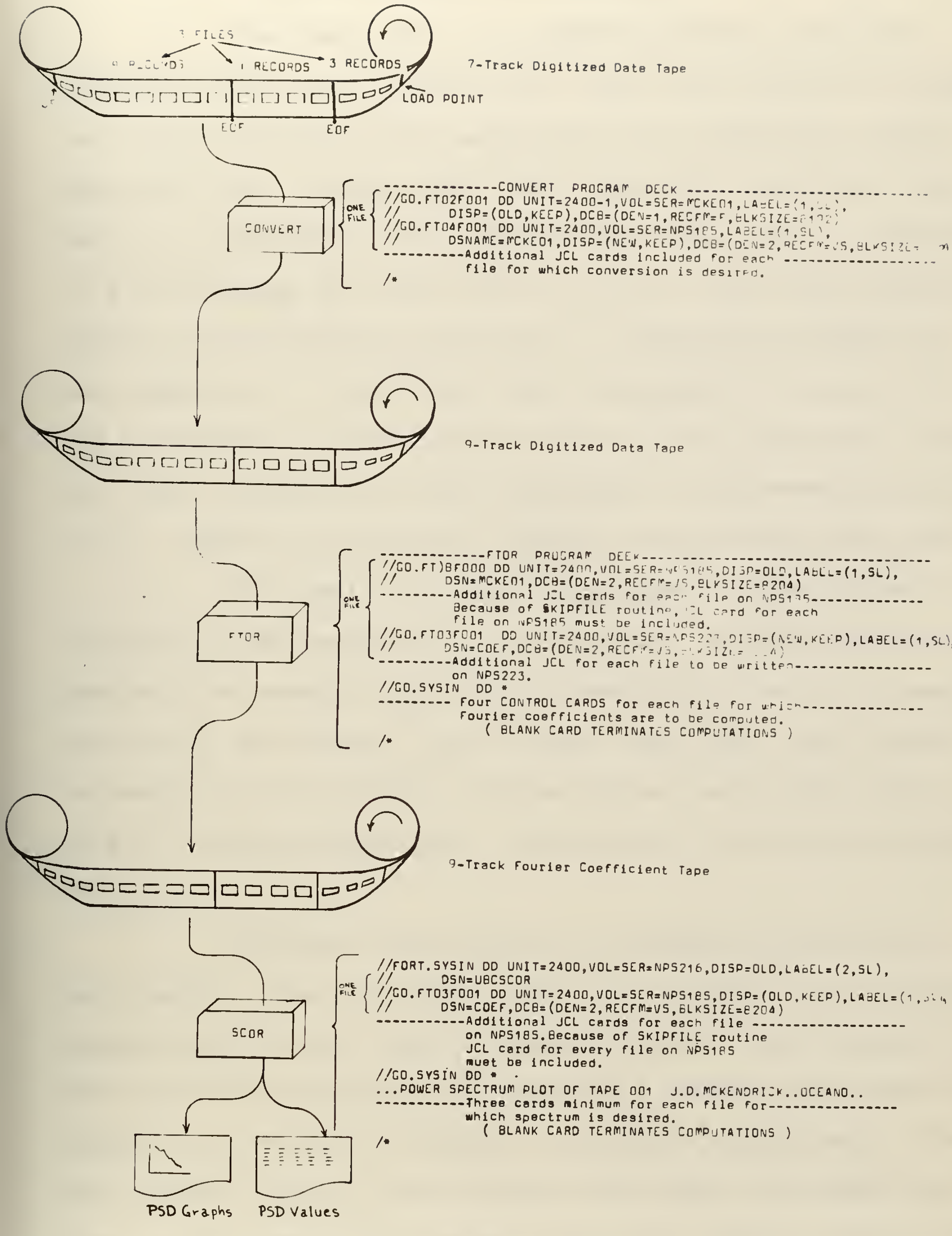


Figure 26. Program Sequencing and JCL Cards Needed for PSD Analysis

a. Program Submission Submission

The best time for processing digital tapes and debugging particular problems was found to be during the summer and winter break when the work load at the computing center was very light. During that time, generally, about ten runs per day, depending on the length of the JOB, could be achieved. If all was going well only one run per day was often adequate; however, when problems developed, an immediate debugging run was essential. It was found that oversights on the part of the programmer, in the form of errors in JCL and control cards, the "dumb" computer was only doing what it had been instructed to do. However, occassional computer malfunctions did occur.

The next best time for program processing was on weekends at the beginning of the quarter. As the quarter progressed, the weekend would offer several runs per day. Toward the end of the quarter, due to heavy work load, only one run at around 3 A.M. resulted, if the job had been input prior to 10 A.M. of the previous day. In general, turn around averaged 24 hours.

The most CPU used for any program sequence was ten minutes. Due to the fact that each analysis uses different parameters, the only rule of thumb which was found useful was that the FTOR procedure took about 22.5 seconds per record (2048 samples per record) to compute the Fourier coefficients. Thus, 400 records required 490 seconds of CPU time. The FTOR program was the most time consuming, thus this figure (22.5 seconds) can be used as an upper limit in estimating CPU time for other programs.

b. Stacking Programs

Another technique used to affect faster PSD analysis involved submitting CONVERT, FTOR and SCOR under one JOB card. The reason the programs were "stacked" was to permit CONVERT, FTOR and SCOR programs to run sequentially. Otherwise, it was necessary to get the results of each program before the next program could be input. If no mistakes were made in the JCL cards and if no problems existed with the tapes, the complete analysis would be made in one run. If a problem in any step was encountered, corrective action was taken and the remaining programs were run individually. The option as to when to stack the programs and when to run them individually varied with the number of files on the tapes. The higher the number of files, the more JCL and control cards required, and the greater the chance for errors.

IV. EXPERIMENTAL PROCEDURE

The initial plan for the investigation of the noise problem, involved digitization and spectral-analysis of real signals. These were to be sine waves, square waves, ramps, and random noise. Since these input-signal characteristics were readily known, the Fourier-transform and, the power sectra were known functions. The first signal, a 10HZ sine wave, did not give the expected spectral values. The digital program was checked to see if its calculations were valid and then the A/D procedure was checked. For purposes of clarity, a chronological discussion of the experimental procedure will follow.

A. ANALYSIS OF PURE SIGNALS

Though the first series of 10HZ sine waves showed an expected energy peak at 10HZ, the exponential decrease of energy with increasing frequency was not expected for a sine function. An assessment was made that the problem could be either in the actual A/D step or in the spectral-analysis programs. Before definite conclusions concerning this problem and the noise sources could be made, it was necessary to gather baseline information on computer analysis of theoretically pure signals.

1. Computer Generated Digital Sine Function

The program listed in Appendix I was used to generate a simulated digital sine signal and the computed samples

were stored on a digital nine-track tape. The format of the tape was made to be compatable for input of this data to the FTOR program. In the sine generation program, the peak voltage could be varied by changing the sine amplitude, and the the sampling rate could be changed by varying Δt. A standard block size of 2048 samples per block was maintained throughout this study. The CONVERT procedure was not needed because the data was already in a hexidecimal format.

2. <u>Computer-Generated Digital, Random Signal</u>

The next step was to test the FTOR and SCOR spectral density analysis of a signal with a wide range of frequencies. To do this, a Gaussian signal, which has a flat spectral density function, was used. The program listed in Appendix II generated a simulated digital random signal by "calling" the computer sub-routing RANDU. The peak voltage values could be changed by altering the constant multiplier of YFL; the sampling rate could be changed by altering the sampling rate specified on the FTOR input data deck. The peak amplitude was maintained at 10 volts; however, the different sampling rates were investigated. Since the data was stored on a nine-track tape with a format compatible with FTOR, the CONVERT program was not used.

B. A/D CONVERSION AND PSD OF LABORATORY SIGNALS

Once characteristics had been established for computer processing of pure control signals, the next step was to digitize actual signals. It was decided that a random or Gaussian signal would give all the information required, and

thus, the sine, ramp, and square wave signals could be by-passed. The random signal was expected to give optimum power-spectral density information for purposes of the study.

1. Random Signal

a. Single Channel Digitization

An Elgenco model 603 A, Gaussian noise generator was used to give a random noise output. Its characteristics are listed in Appendix III. A frequency setting of 5Hz to 20 KHz, attenuation schale X1.0, output voltage reading 2.62 Vrms was input to a Khron-Hite filter model 3321 set at 2000 Hz, low pass max. flat and Odb gain. The filter output was input to the Hybrid computer through an operational amplifier with a ten volt gain. A sampling rate of 5000 SPS was selected. A total of 41 seconds of the signal was digitized onto a seven-track tape. An End-of-File mark was written onto the tape by typing the EOF option on the teletype keyboard.

b. Dual Channel Digitization

The second file on the tape mentioned above was filled with data from two input channels which were digitized simultaneously. The Elgenco noise generator described above was used as one input, and the other input was from the random-noise generator which is built into the Ci 5000.

The noise-generator was utilized with the same settings as above; however, the filter was reset to 1880Hz. The random-noise generator from the Ci 5000 was input to a Krohn-Hite filter also set at 1880 Hz. The output of this filter was fed into a different operational amplifier with

a gain of 10. The sampling rate was lowered to 4000 SPS. A total of 25.5 seconds of signal was digitized. Though the single and dual channel cases both considered 100 records, this shorter digitizing time occurred because, two samples were being taken, simultaneously, at the rate of 4000 SPS:

$$\frac{2048 \frac{\text{Samp}}{\text{Blk}} \times 100 \text{ Blks}}{4000 \frac{\text{Samp}}{\text{Sec}} \times 2} = 25.5 \text{ Sec.}$$

An End-of-File mark was again written on the tape to end this file of data.

The CONVERT program as used to convert from seven- to nine-track tape. The program SCOR allowed the computation and plotting of the spectrum of each channel and the co-spectrum and the quad-spectrum of one channel with another.

2. Random Signal and Sine Signal

To determine whether a sine wave could be picked out of the random noise, a 1000 Hz sine and random signal were digitized. An attempt to digitize a single channel of the sine combined with the random signal failed due to a faulty patch on the analog board. PSD values were obtained, seemingly because the open amplifier actually picked up stray signal. The sine amplitude was increased and the a second file was digitized. The signals in these two files were digitized at 5000 SPS and filtered at 2000 Hz.

Dual channel digitized samples of the 1000 Hz sine with amplitude ±20 volts and Gaussian signals filled the third file. The amplitued was increased to ±30 volts and the two separate sine and Gassian signals digitized into the

fourth file. The signals in these third and fourth files were digitized at 4000 SPS and filtered at 2000 HZ.

C. DATA FROM GEOPHYSICAL SIGNALS

Atomspheric-temerature and velocity signals have previously been recorded on one-inch magnetic tape by Boston [Ref. 1]. These signals were used as a final check on the system to determine if correct spectral values could be achieved.

The signals were reproduced on a Sangano Model 3562 FM tape recorder at 60 ips. The tape playback output was filtered at 1000 Hz for the temperature signal, and at 2000 Hz for velocity, the differentiated velocity and temperature signals. The sampling rate for the temperature-signal digitization was 2000 SPS and the other three signals were sampled at 4000 SPS. The filter setting was low-pass max, flat, Odb gain.

V. ANALYSIS OF RESULTS

A. PSD OF COMPUTER GENERATED SIGNALS

1. Sine Wave

Figure 27 was the spectral plot of a computer-generated sine wave. The PSD values were computed for 24 records of signal giving a total signal length of 11.9 seconds. The total integrated power was .499 V^2; However, the power in the 8.12 Hz band centered about 7.11 Hz had a total of .495 V^2 (8.13 Hz X 6.09 X $10^{-2} V^2$/Hz). Essentially, all the power was contained in the band between 2.05 Hz and 11.17 Hz. Though this appeared to be a wide band, the whole region from 11.7 Hz to 263.1 Hz had less than .8 percent of the total power:

$$\frac{.499 - .495}{.499} \times 100 \sim .8\%$$

Another test run with a 200 Hz sine wave with a peak-to-peak amplitude of 10 volts, proved inconclusive due to an error in specifying the sampling rate on the FTOR data-card input. A sampling rate of 400 SPS was specified, rather than 500 SPS, which was the actual rate used. The expected total power value of 50.V^2 was achieved; however, the error with the sampling rate shifted the frequency peak from 200 Hz to 159 Hz. Though the error produced erroneous results, the effect of not specifying the correct sampling rate was observed.

2. Gaussian Noise

Figure 28 was the PSD plot of the computer-generated random signal. The digital samples of the signal simulated

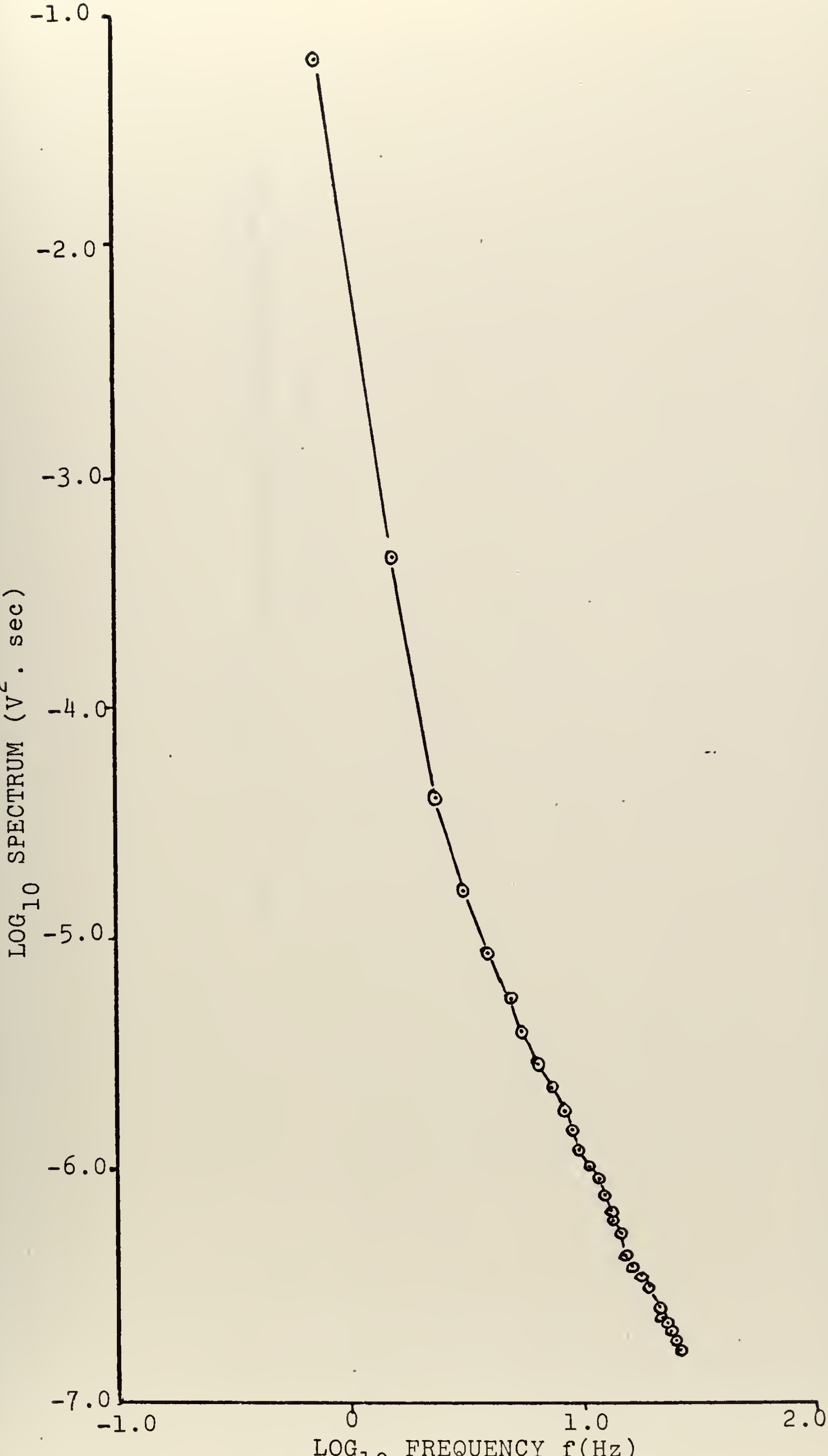


Figure 27. PSD Plot obtained from Computer Generated 10 Hz Sine Wave

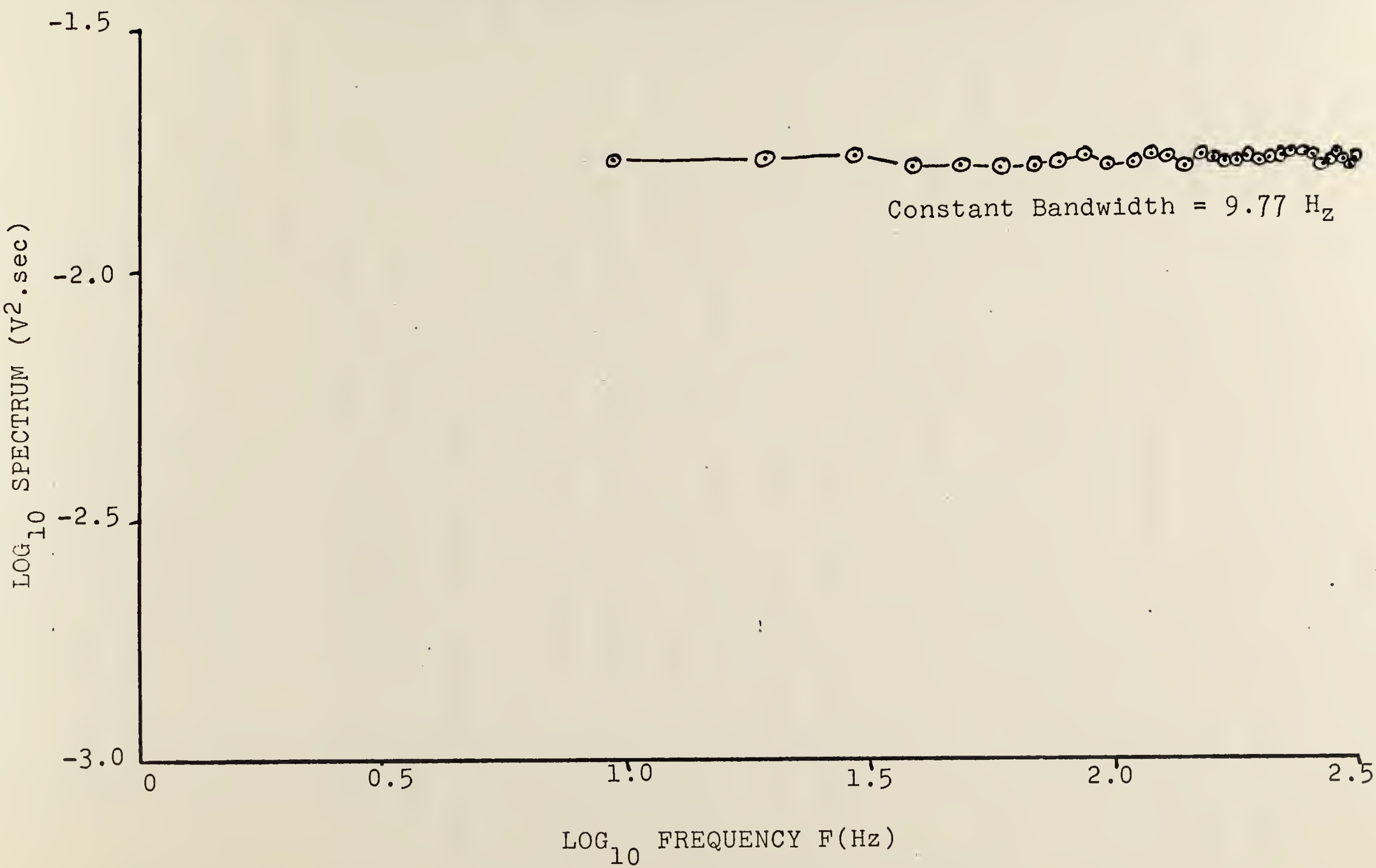


Figure 28. PSD Plot of Computer Generated Random Signal

a random signal of 9.8 seconds in length, sampled at 5000 SPS. The spectral level was found to be very flat with increasing frequency. Average spectral density of 3.30×10^{-3} V^2 /Hz varied from a high value of 3.45×10^{-3} V^2/Hz to a low value of 3.25×10^{-3} V^2/Hz. Actual variance was very low and quite uniform. No frequency spikes were observed and the conclusion was drawn that the random number generating sub-routine RANDU produced a true random series for at least the first 50,000 numbers. It was noted that the random-number generating capacity of this sub-routine is 2^{39} numbers before the series repeats itself. Since only 2^{17} numbers were used, the full potential of the number generator was not fully tested.

Figure 29 is the SPD plot of the same random series sampled at 1000 SPS rather than 5000 SPS. This changed the record length to 2.05 seconds, and since the results for 100 records was computed, the total length of signal sampled was 205 seconds. Although the sampling rate-change produced a higher PSD value, the mean showed little variation. The mean level was about 1.57×10^{-2} V^2/Hz with a low of 1.62×10^{-2} V^2/Hz and a high of 1.72 V^2/Hz. The variance around the mean level was very small. As expected, averaging over a fewer number of records (24 versus 100), the variance was higher.

B. PSD LABORATORY SIGNALS

1. <u>Single Channel Sine</u>

a. Signal Leakage Into Open Amplifier

Figure 30 was the PSD plot of the spectrum of signals with peak voltages of ±2.0 and ±3.0. The signals

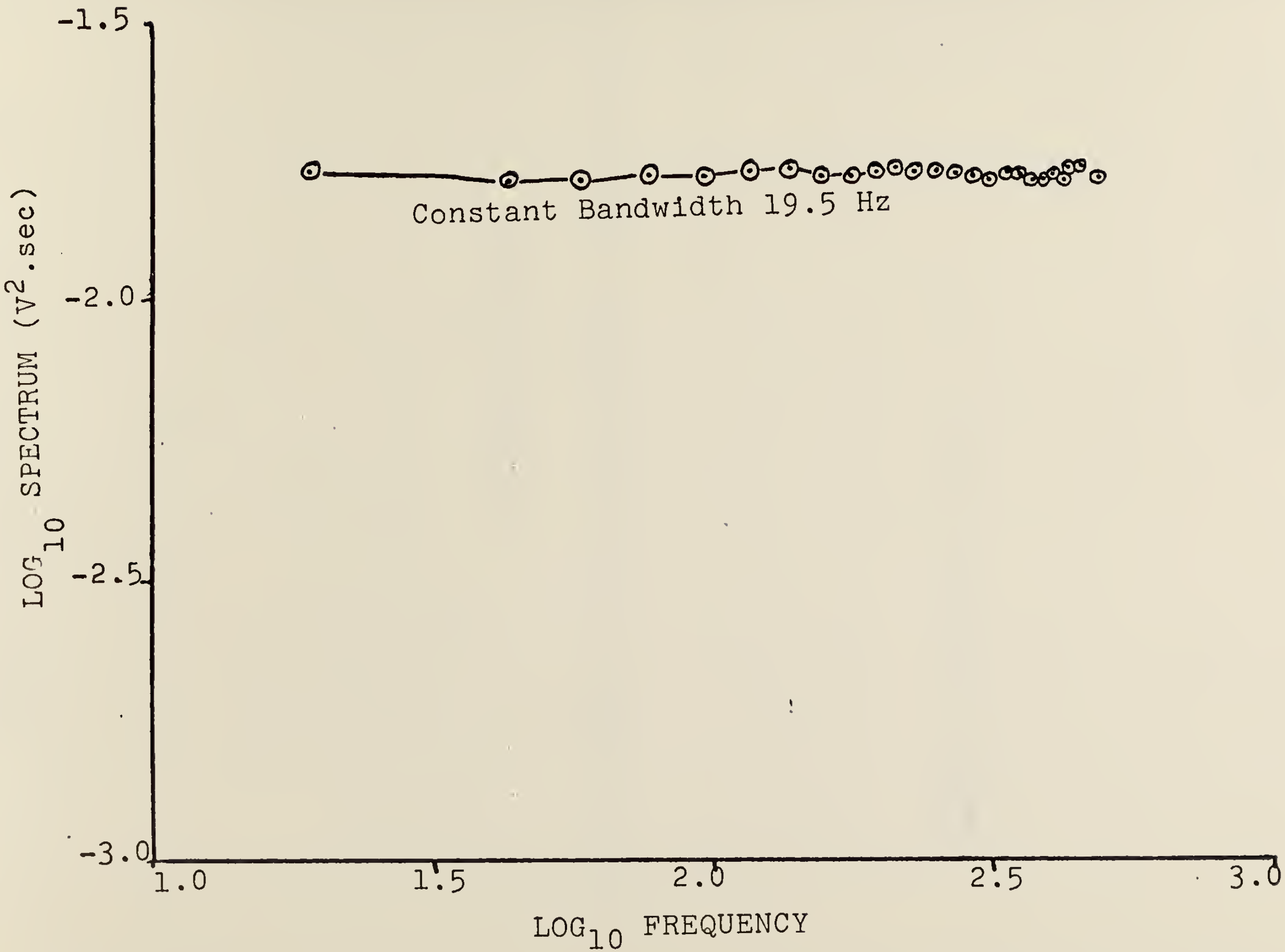


Figure 29. Spectral plot of Computer Generated Random Signal Peak Amplitude 10 Volts, Record Length = 2.05 Seconds, Sample Rate = 1000 SPS

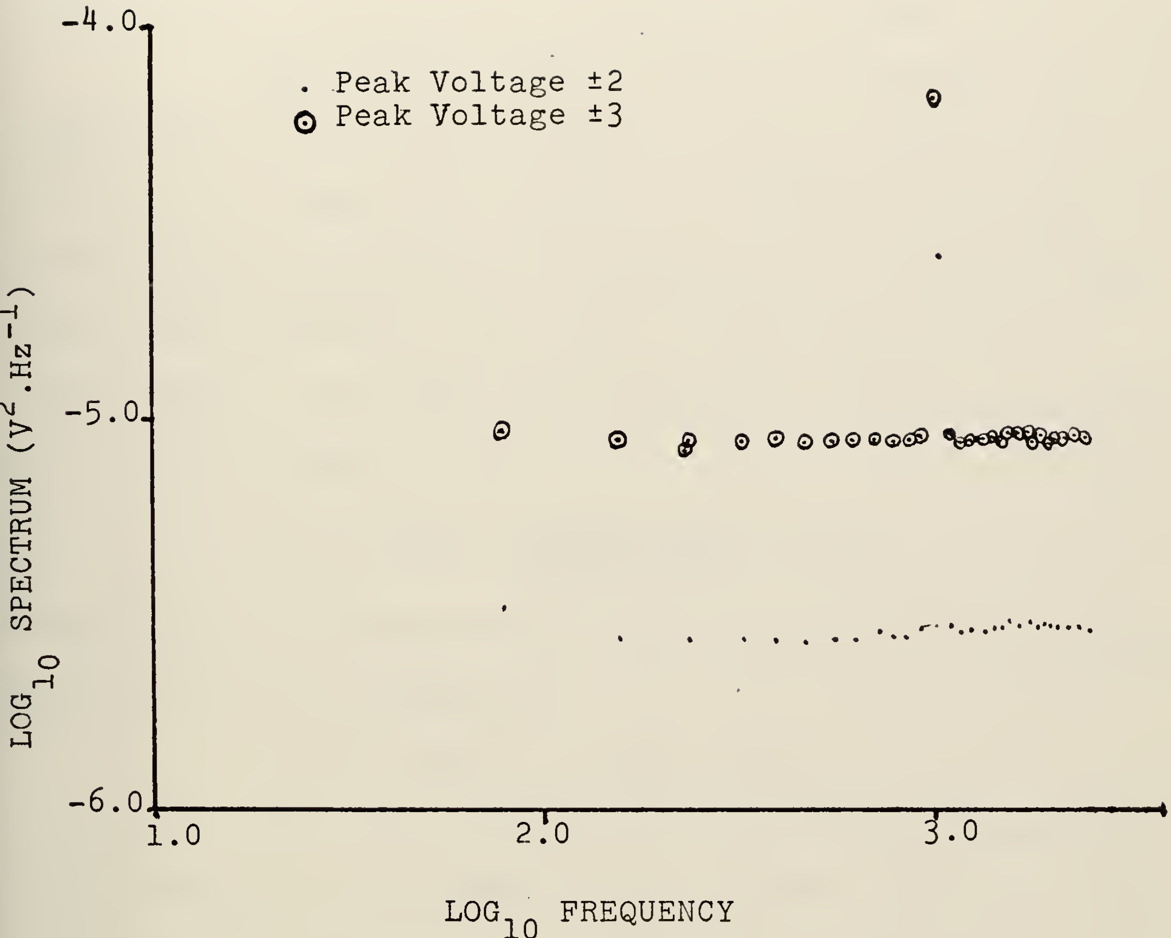


Figure 30. Signal Leakage into Open Amplifier

were picked up by an open input amplifier, which was next to the amplifier into which the signals were actually input. The open amplifier, whose output was being digitized, acted like an antenna in picking up these stray signals. The plots show considerable consistency and several conclusions can be made.

The spectral peak of the ±3 volt signal was higher than that of the ±2 volt signals. The peak PSD was 2.68 X 10^{-5} V^2/Hz for the lower and 6.58 X 10^{-5} V^2/Hz for the higher signal. These values multiplied by the band width, 78.1 Hz in both cases, gave power of 2.09 X 10^{-3} V^2 and 5.12 X 10^{-3} V^2 respectively. The power ratio produced by this difference in output voltage was

$$\text{Power Ratio} = \frac{6.56 \times 10^{-5}}{2.68 \times 10^{-5}} = 2.45.$$

Since this was a power ratio, the voltage ratio is the square root of the power ration, or

$$\text{Voltage Ratio} = 2.45 = 1.56$$

The actual voltage input into the open amplifier was computed from the power of each signal. The lower power 2.09 X $10^{-3}V^2$ resulted from an input voltage of ±4.5 X 10^{-2}V, and 5.12 X $10^{-3}V^2$ resulted from an input of ±7.2 X 10^{-2}V.

The expected voltage ratio was computed using the observed input voltage values:

$$\text{Voltage Ratio} = \frac{3.0}{2.0} = 1.5$$

The power ratio was;

$$\text{Power Ratio} = (1.5)^2 = 2.25$$

The ratio of the signal voltage in the input line to the actual computed signal voltage gave the actual percent of signal picked up by the open amplifier.

$$\frac{4.6 \text{ X } 10^{-2}}{2.0} \text{ X } 100 = 2.3\% \text{ for the 2V signal}$$

$$\frac{7.2 \text{ X } 10^{-2}}{3.0} \text{ X } 100 = 2.4\% \text{ for the 3V signal}$$

Assuming the leakage was coming from the voltages in the input lead to the open amplifier, the percent of signal leakage would be the ratio of the voltage in the input lead to the signal voltage computed from the PSD. For the ±2 V and ±3 V signals the percent of signal picked up by the open amplifier was 2.3 percent and 2.4 percent respectively:

$$\frac{4.6 \text{ X } 10^{-2}}{2.0} \text{ X } 100 = 2.3\%$$

$$\frac{7.2 \text{ X } 10^{-2}}{3.0} \text{ X } 100 = 2.4\%$$

If the signal was leaking from the closed amplifier, the leakage was .2 percent for both cases.

$$\frac{4.6 \text{ X } 10^{-2}}{2.0} \text{ X } 100 = .2\%$$

$$\frac{7.2 \text{ X } 10^{-2}}{3.0} \text{ X } 100 = .2\%$$

Thus, only a small signal leakage was observed and it was independent of signal amplitude.

b. Effect of Increasing Signal Amplitude

Figure 31 is a PSD plot showing the effect of increasing the voltage of the signal going into the data taking amplifier. The input signals had peak-to-peak voltages of ±20 volts and ±30 volts.

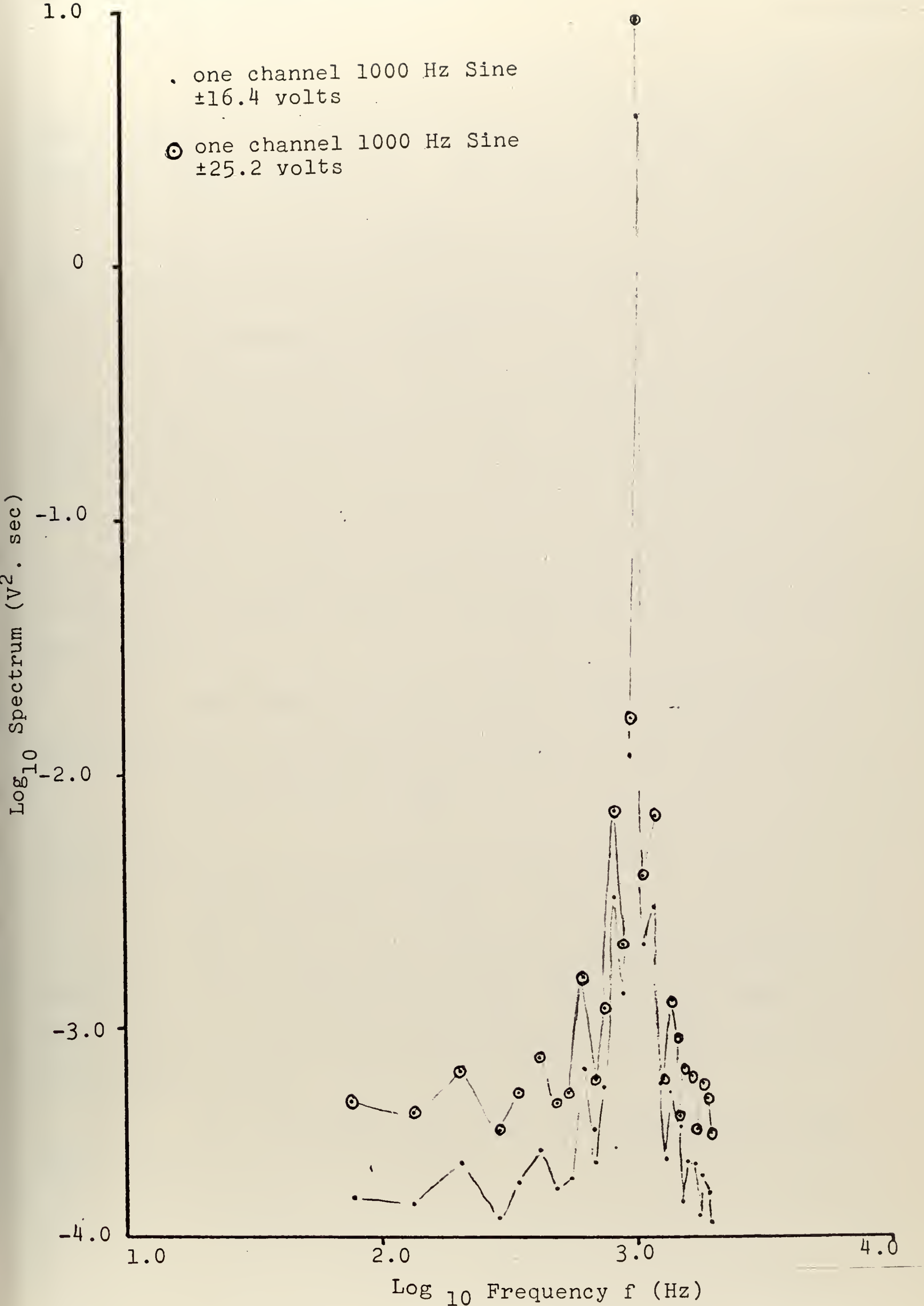


Figure 31. Effect of Increasing Amplitude on PSD Plots of Real Sine Signals

The PSD for the peaks 3.92 V^2/Hz and 9.29 V^2/Hz were computed for a bandwith of 68.4 Hz. The power was 268V^2 and 635V^2 respectively. This gave a power ratio of:

$$\text{Power Ratio} = \frac{9.29}{3.92} = 2.37$$

The voltage ratio is the square root of the power ratio or

$$\text{Voltage Ratio} = \sqrt{2.37} = 1.54$$

Using the power spectral density to compute the voltage ratio, 268V^2 implied an input of 16.4 V_{rms} and 635 V^2 implied an input of 25.2 V_{rms}. This implied peak-to peak voltages of ±23.2V and ±35.6V. Due to the inaccuracy involved in reading peak-to-peak voltages from the oscilloscope on the Ci 5000, the observed inputs of ±20V and ±30V could have been ±5V in error.

Assuming, the inputs were of ±20V and ±30V, the expected voltage ratio would have been:

$$\text{Voltage Ratio} = \frac{30}{20} = 1.5$$

and the expected power ratio would have been:

$$\text{Power Ration} = (1.50)^2 = 2.25$$

The observed and computed power ratios compared favorable, and the difference between ovserved and computed peak-to-peak voltages (20V Vs. ±23.2V and ±35.6V) were within acceptable limits.

The theoretical power for sine waves of ±20V and ±30V was found from the formula:

$$P = \frac{V^2}{2}$$

where V is peak-to-peak voltage. This gave power of 200 V^2 and $450V^2$ for the two voltage signals respectively. The difference between expected power level and the power level derived from the spectral plots was assumed to be due to the error in reading the input signal amplitudes.

2. <u>Single Channel Gaussian Signal</u>

Figure 32 was the spectral plot of 40.06 seconds of a random signal sampled at 5,000 SPS. The spectrum level was very flat to about 1.5 KHZ. Beyond 1.5 KHz, rapid decrease in the power spectral density with increasing frequency was to be expected. The 3db down point occurred at 2.0 KHZ. The slope of the filter, as specified in the equipment characteristics, was -48db/octave. The observed slope was very close to -96db/octave. This value would be expected if two filters had been cascaded, but this was not the case. The spectrum was quite free of noise spikes and had no 60 Hz harmonics present. If 60 Hz noise was present, its level was well below -17.5db/Hz.

The spectral level of 1.73 X $10^{-2}V^2$/Hz compared favorably with the input signal. Specifications for the Elgenco noise generator gave a spectral density of approximately 5 X 10^{-3}V/ Hz at $1V_{rms}$. The input signal had a meter reading of $2.62V_{rms}$. The gain factor was 10. The computed spectral density was 2.69 X $10^{-2}V^2$/Hz.

$$\left(\frac{5 \times 10^{-3} V}{Hz}\right)^2 \quad (2.6)^2(10)^2 = 2.69 \times 10^{-2} V^2/Hz$$

Since no accurate spectral density information on the noise generation was available, these values are considered to compare favorably.

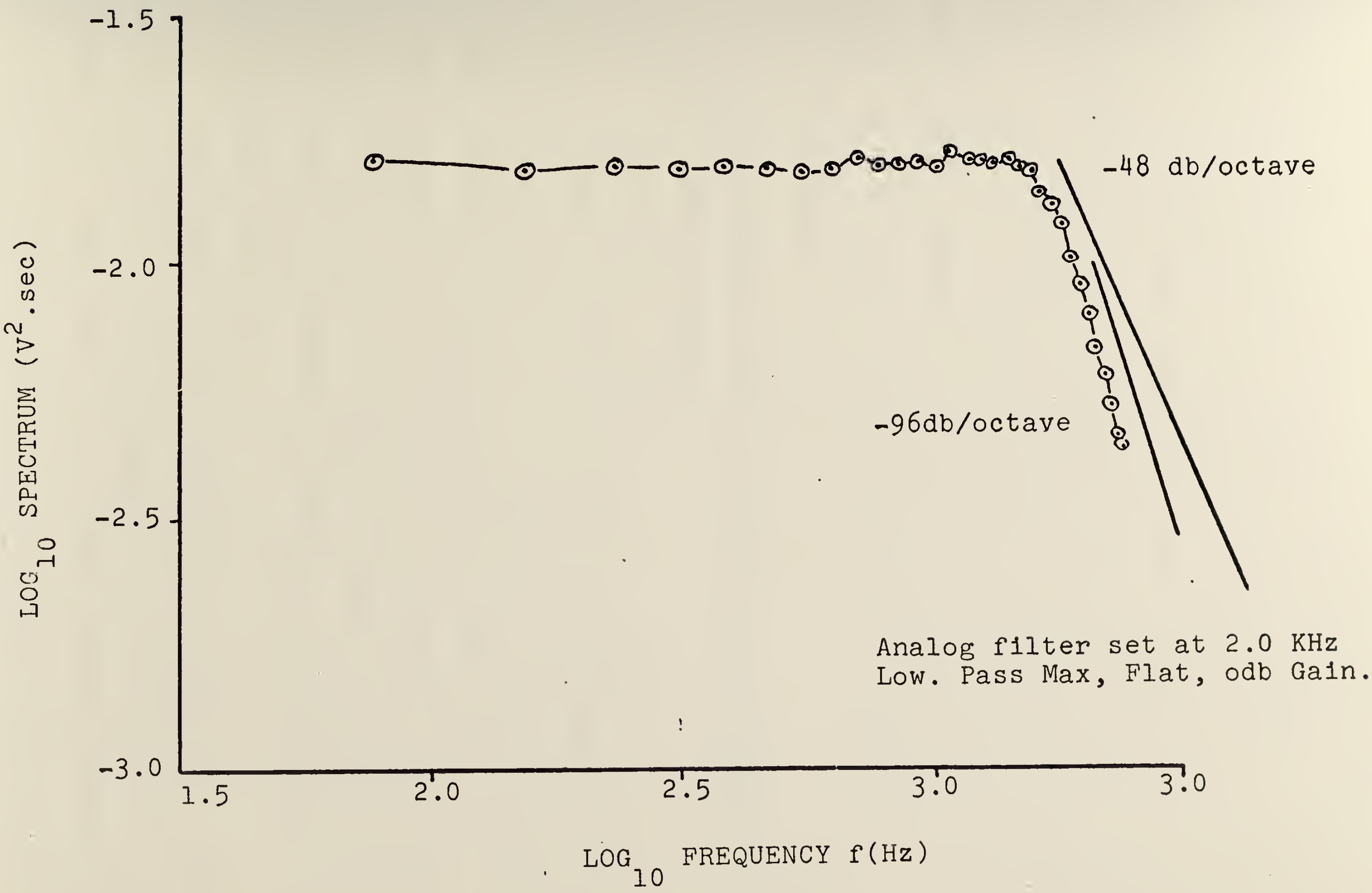


Figure 32. Spectrum of Gaussian Signal

3. Two Channel PSD of Gaussian Signals

Figure 33 is the PSD plot of one of two channels of random noise which were digitized simultaneously. One spectrum resulted from the Elgenco Gaussian noise generator and the other spectrum resulted from the high frequency random-noise generator of the Hybrid computer. The flat spectrum of the Elgenco generator was contrasted by the very "colored" spectrum of the Ci 5000 random-noise generator, whose low frequencies contain much power. Because of its deviation from the Gaussian characteristics, future use of the Hybrid noise-generator should be avoided when a Gaussian generator is wanted.

C. PSD OF TURBULENCE SIGNALS

1. General Signal Characteristics Found; Comparison with Previous Results

a. Temperature Signal

Figure 34 showed the PSD of the temperature signal recorded by Boston [Ref.1]. A comparison was made between the PSD valued obtained from 56 seconds of signal that was digitized and analysed at the Naval Postgraduate School Facility, and PSD the values obtained by Boston from the same section of signal that was digitized and analyzed at the University of British Columbia facility.

The general PSD characteristics from both analysis compared favorably in slope magnitude and 60 Hz harmonic peaks. The strong -5/3 slope region was evident on both spectra between 10 Hz and 120 Hz. Significant harmonic levels were

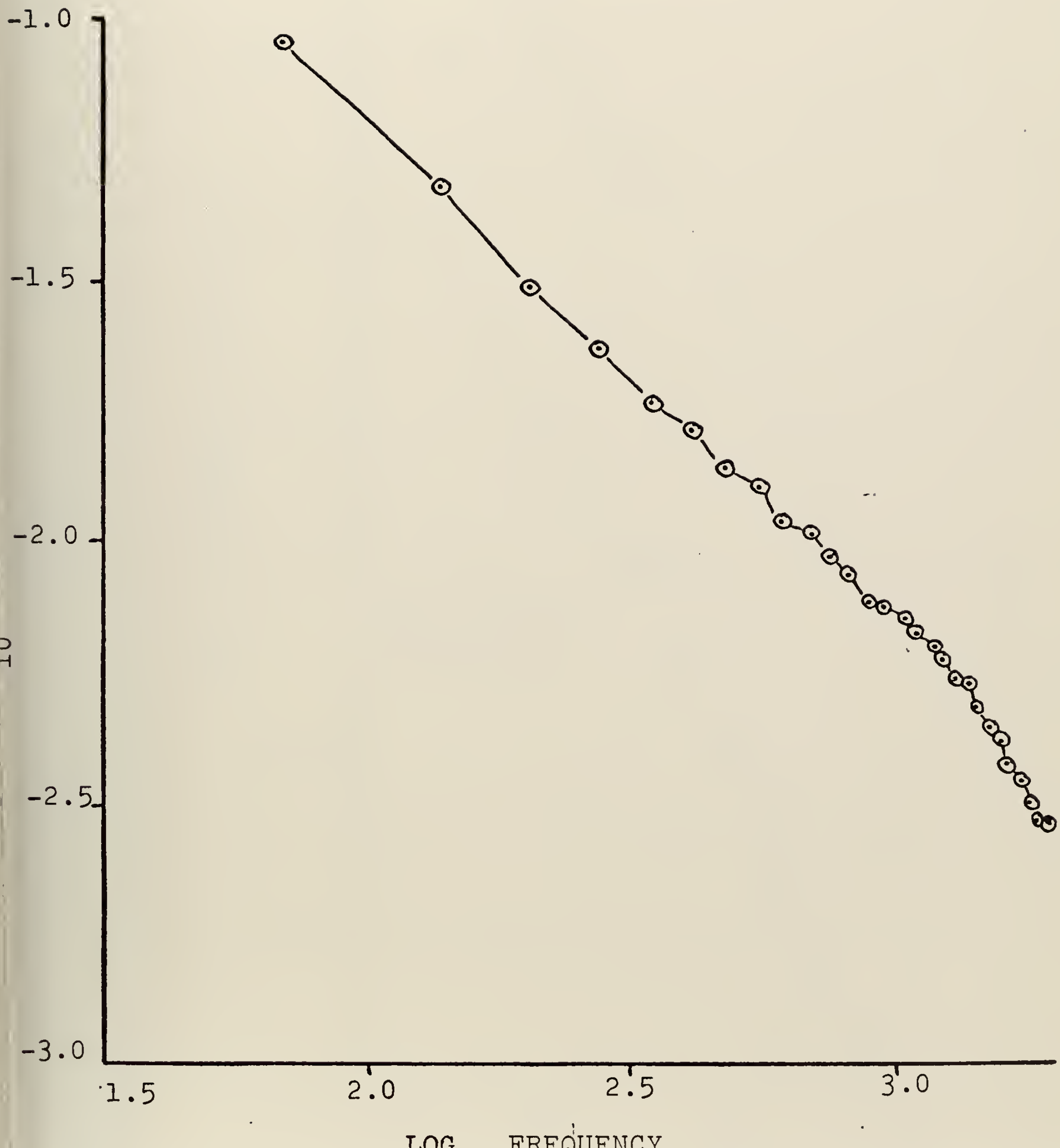


Figure 33. Spectral Plot of Ci 5000 random Noise Generator output. Sampling Rate = 4000 Samp/Sec.

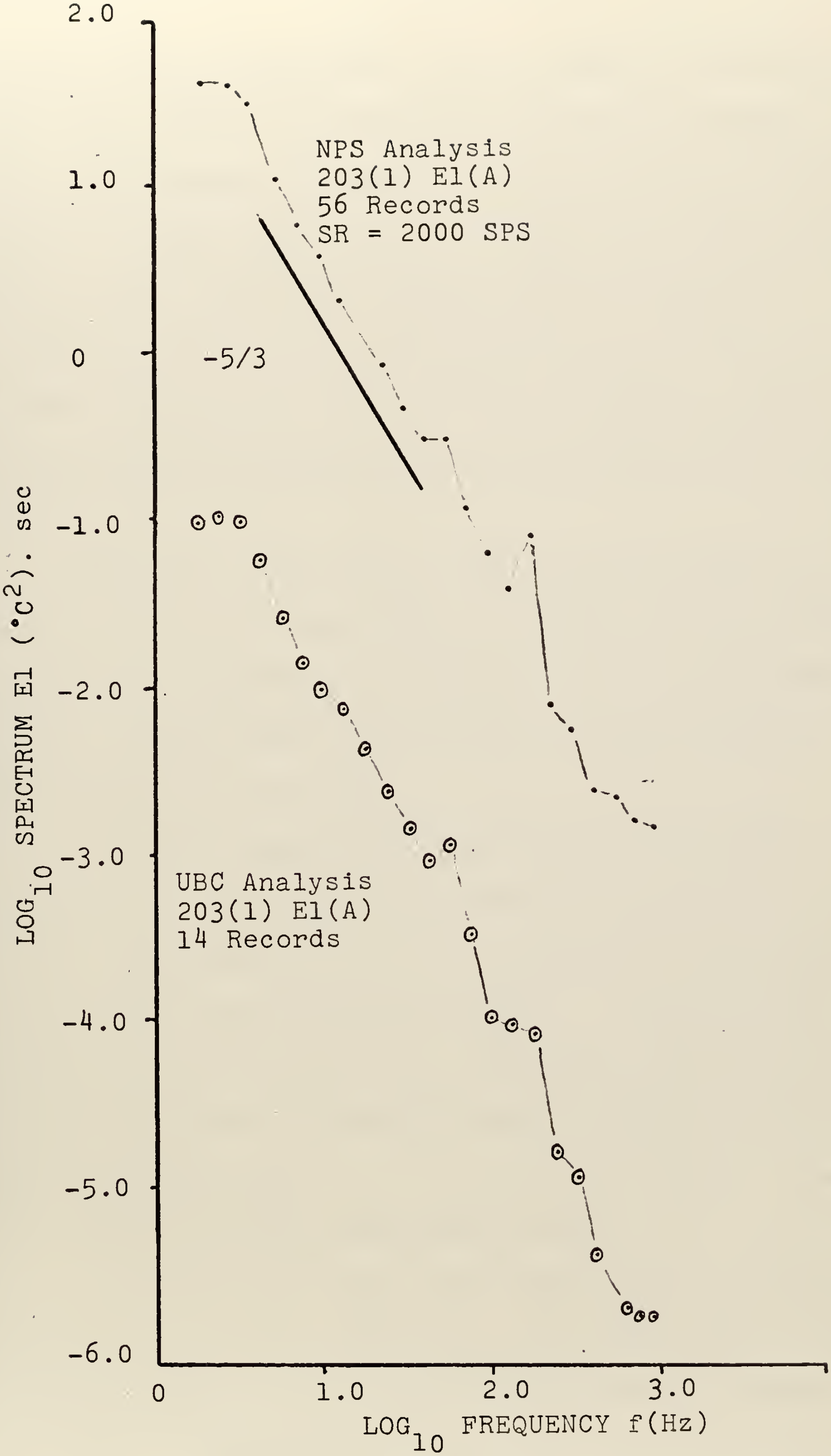


Figure 34. Comparison of Magnitude of Temperature PSD Results Obtained at UBC and at NPS

evident in the Naval Postgraduate School study at the first (60Hz) and third (180Hz) harmonics and a less significant ninth (540 Hz) harmonic. Boston's results showed a significant first harmonic. The third harmonic, though detectable, showed a very minor level; the ninth harmonic was not detected. A slight increase in the slope to approximately -7/3 was observed between 200 Hz and 400 Hz (600 Hz in Boston's results) followed by a marked decrease in the slope between 600 Hz and 1000 Hz.

The major difference between the two spectra is the constant power level difference. The Naval Postgraduate School results were higher by a constant factor of 400 which implied that a voltage gain difference of 20 was present in the Naval Postgraduate analysis.

As can be seen from Figure 35, in which each Naval Postgraduate School PSD value has been reduced by a factor of 400, the spectra for the two analyses compare vary favorably in slope and relative magnitude. A temporal PSD plot described in a later section, would have been very helpful in detecting noise in Boston's results; however, time did not permit its production.

b. Differentiated Temperature Signal

A comparison of short duplicate sections of the differentiated temperature signal was not undertaken in this study. A long section of the signal was, however, analysed. Figure 36 shows the comparison for the PSD from five minutes of signal obtained in this study with the PSD from three

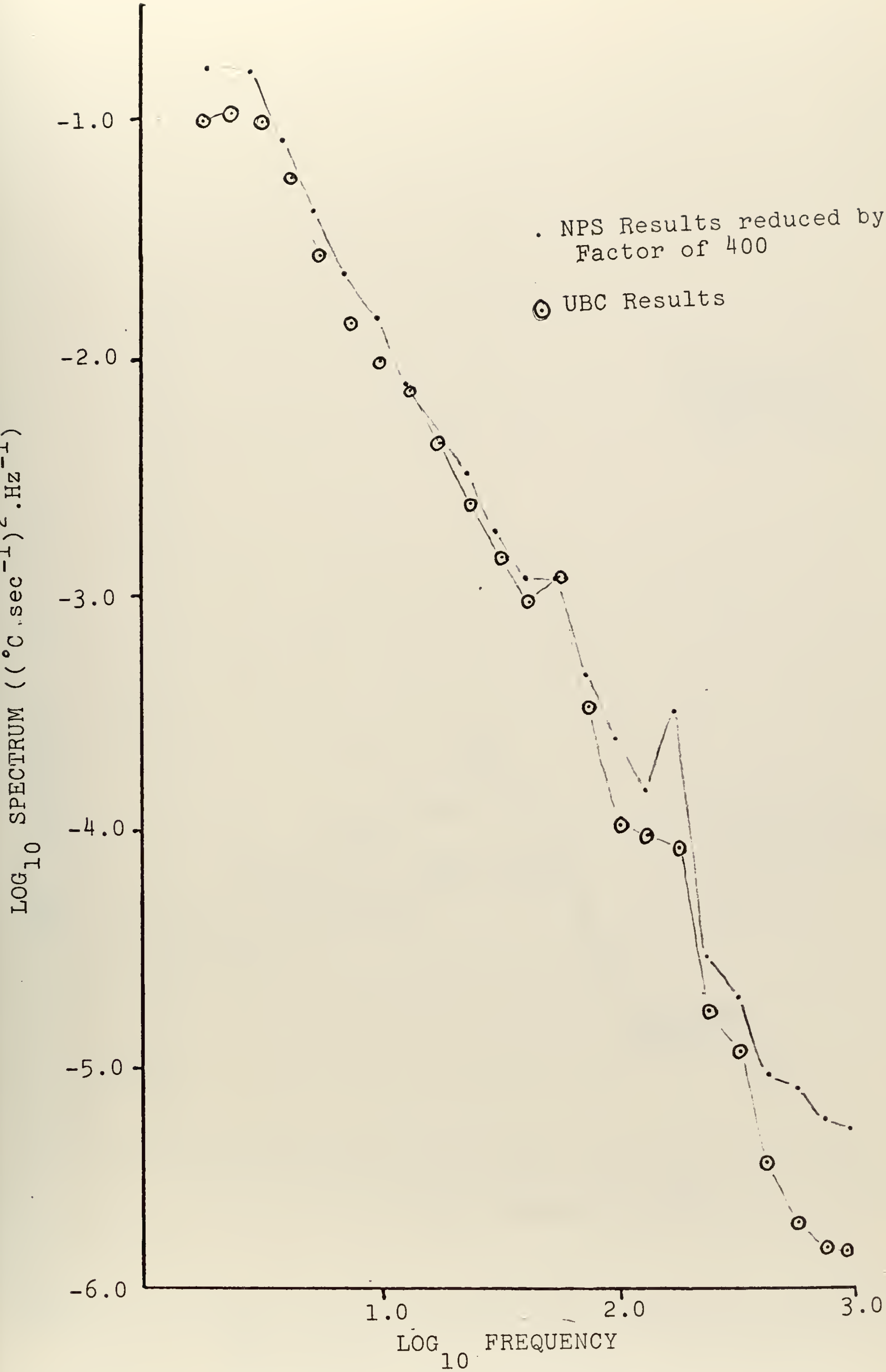


Figure 35. Comparison of Slopes of Temperature PSD. NPS Results Reduced by Factor of 400.

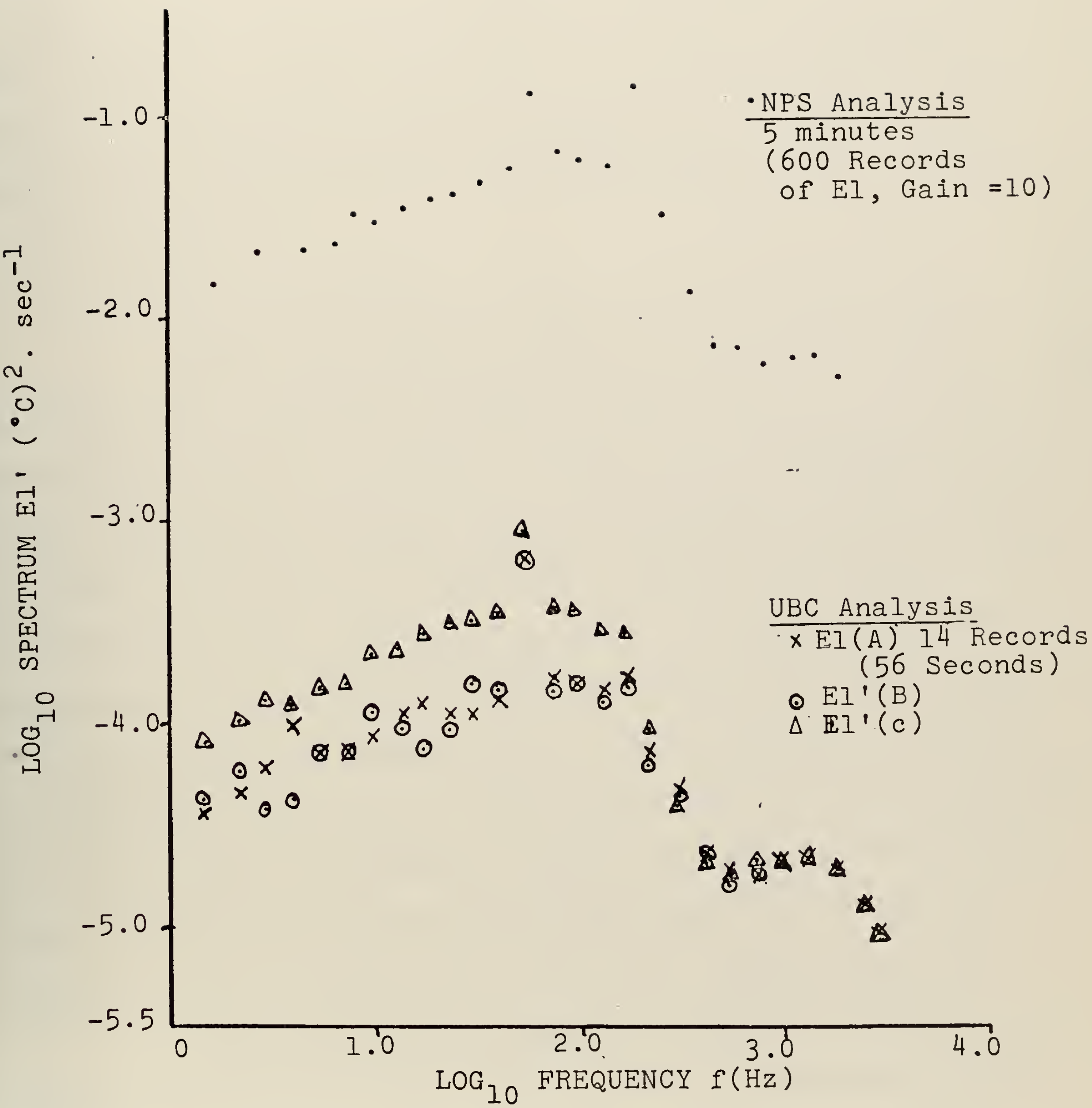


Figure 36. Comparison of NPS Analysis of Differentiated Temperature with UBC Analysis (Exponential Bandwidth Used)

different sections of data obtained by Boston (Ref. 1]. The general characterisitics were observed to agree favorable, though the Naval Postgraduate School results were greater by a constant factor of 300.. In Figure 37 the Naval Postgraduate School results were reduced by a factor of 300 and were found to follow closely the results from E 1'(B) and E 1'(C). The first harmonic of 60 Hz was found in the Naval Postgraduate School and University of British Columbia data; however, the strong third harmonic found in the NPS results wasn't evident in the UBC results.

c. Velocity Signal

Figure 38 shows the PSD of a section from the velocity signal recorded by Boston on tape 203 (1). It compares with a section analysed by Boston and is referred to as U(A). The original analysis was conducted for a sample which was about 56 seconds in length. The original sampling rate was 200 SPS which gave a Nyquist frequency of 1.0KHz. The Naval Postgraduate School analysis was conducted on the same section of signal, using a sampling rate of 4000 SPS and a block size of 2048 samples per block. Since only 56 records were analyzed, the total length of signal was only 51.2 seconds. Though the Naval Postgraduate School results are based on a slightly shorter signal, the PSD plots show quite similar results.

The power level of the University of British Columbia analysis is lower than the Naval Postgraduate School analysis by a factor of 340, which implies a voltage gain difference of 18.4 existed between the two sets of results. Harmonics of 60 Hz were found in both cases in that the first

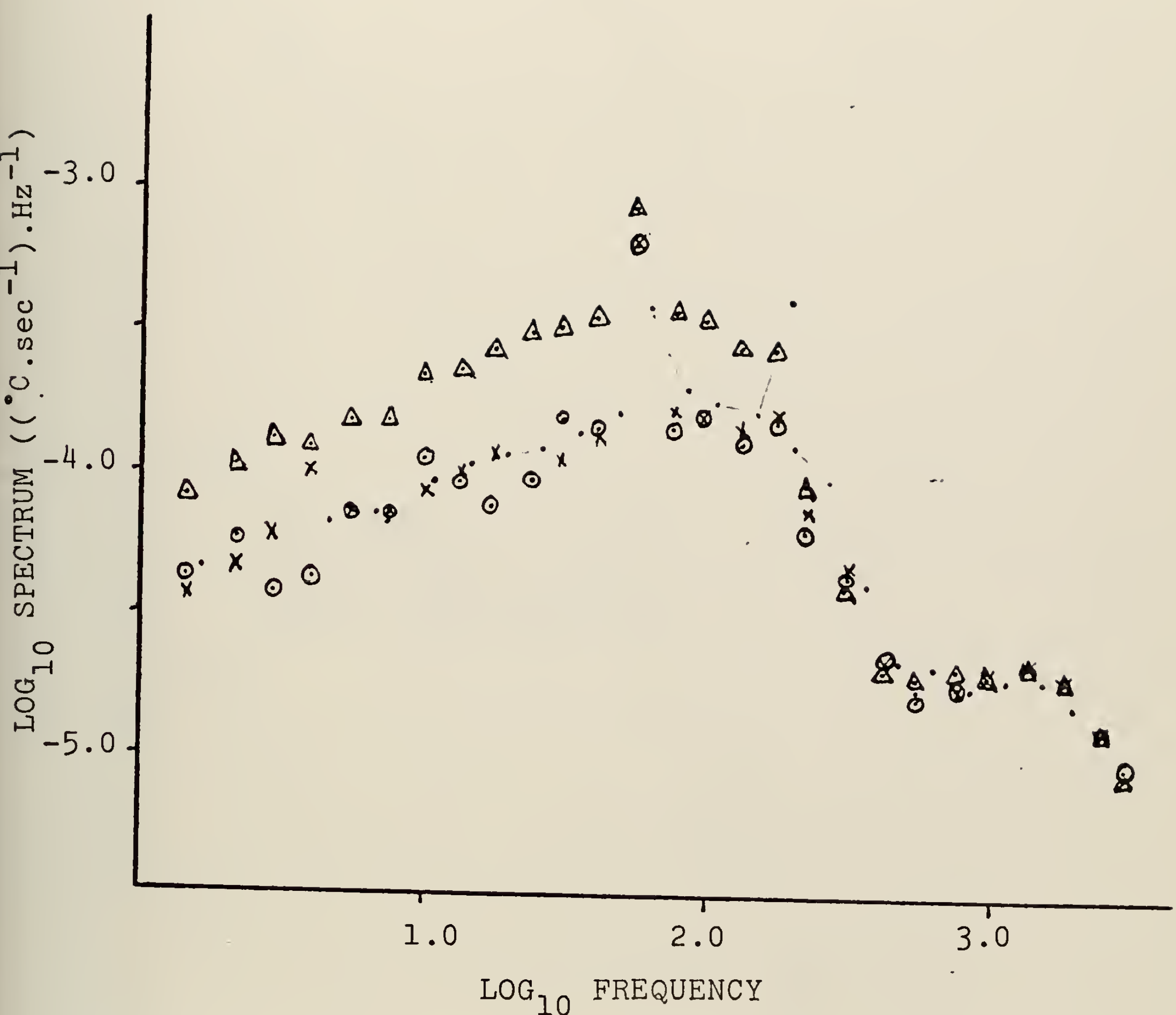


Figure 37. Comparison of UBC and NPS Analysis of Diffrentiated temperature. NPS Analysis reduced by Factor of 300 Exponential Bandwidth

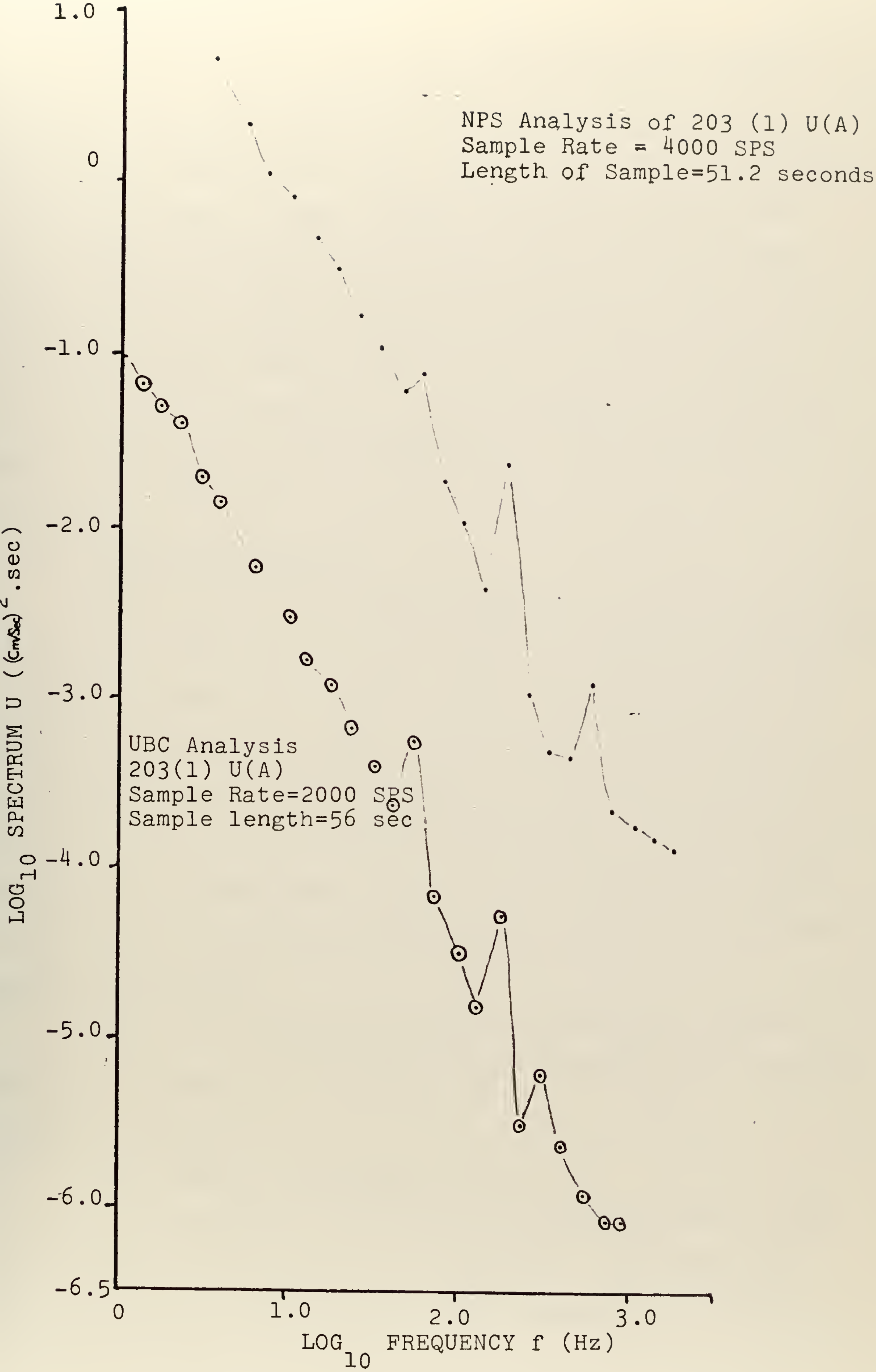


Figure 38. Comparison of Magnitude of Velocity PSD Results Obtained at UBC and NPS.

and third harmonic peaks stood well above the -5/3 slope of the signal. A marked deviation in results occurred in the PSD plots for frequencies higher than 250 Hz. The University of British Columbia results showed a spectral peak at about 320 Hz whereas the Naval Postgraduate School results showed a peak at 620 Hz. Since the Nyquist frequency of this study was 1000 Hz higher than Boston's results for U(A), previous values had not been reported for the frequency range 1000 to 2000 Hz. Indications from Figure 39 are that the slope has decreased from the -5/3 value to a value of -2.5/3.

2. New Results Obtained

Based on the encouraging comparison of results obtained in this study with the results obtained by Boston at University of British Columbia, the values obtained for the longer time period of five minutes appear to offer new insight into the statistical properties of the geophysical processes measured. Generally, the results obtained indicate that the statistical properties, of judiciously chosen short sections of a signal, do give valid indications of the nature of over-all processes under consideration.

With the increased capability afforded by computer analysis of data, new insights may be achieved toward viewing natural phenomena.

a. Temporal Variations in the PSD

A different, but not by no means new way of displaying computer PSD values of geophysical processes, is through temporal PSD analysis. This technique allows the

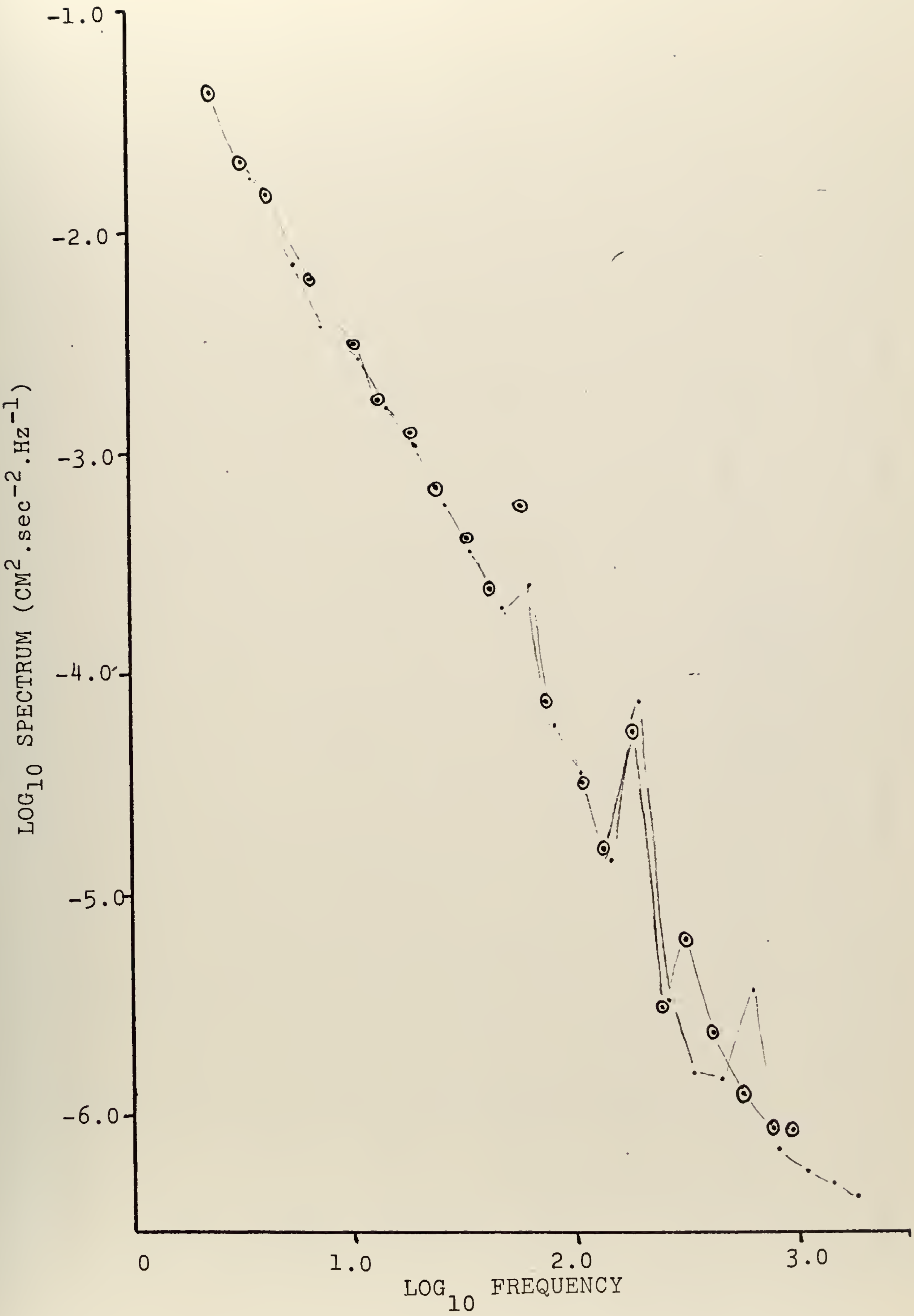


Figure 39. Comparison of Slopes of Velocity PSD NPS Results Reduced by Factor of 340

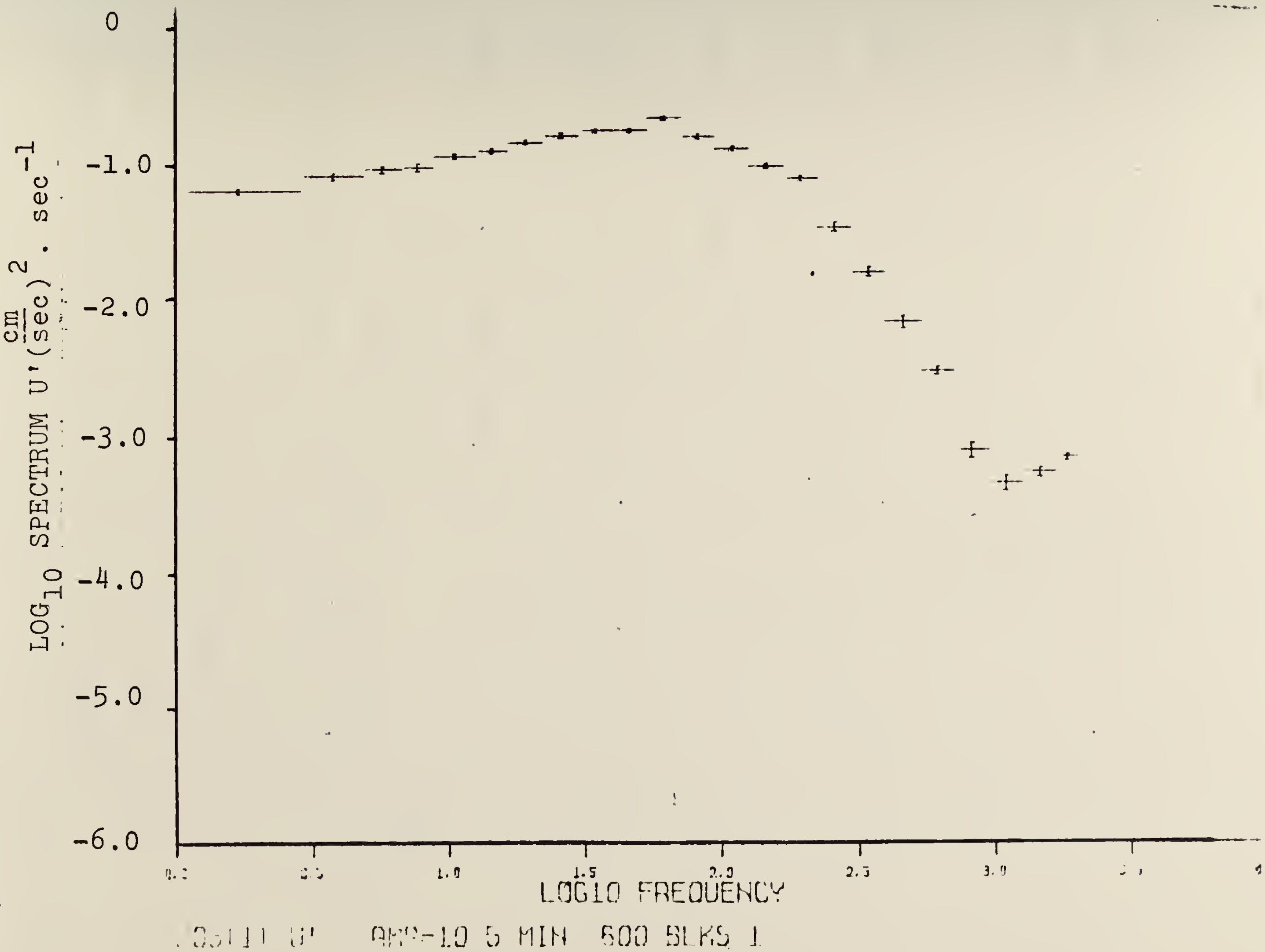


Figure 40. NPS Analysis of Differentiated Velocity Signal: Boston 203(1) U'

investigator the opportunity to relate fluctuations of the PSD with signal fluctuations. A section of a signal which appeared to have a nominal signal amplitude, as in Figure 41, had a low PSD value (as seen in the section of PSD plot marked 10.24 in Figure 42. Though the characteristics of the PSD are indeed determined by sampling rate, the number of samples in each digitized record, filter settings, background noise, etc., meaningful results can be achieved within these limitations.

Since the identity of each digitized record (digitized block of data) was maintained throughout the analysis under UBC FTOR and UBC SCOR, the PSD of sequential sections of the signal could be computed. Temporal variations in the PSD of temperature and velocity signals were drawn from the results obtained by sequentially analyzing a constant number of records. Plots were also drawn from the results of analyzing an increasing number of records, beginning with the first record. These plots showed that the fluctuating PSD became more stationary when more and more data is analyzed.

(1) Temperature.

Figure 42 shows the temporal variation of the temperature PSD. A total of 300 records were analyzed to give a total time of 307.2 seconds. To compute the PSD values, thirty passes were made through the Fourier coefficients with the PSD being computed for each set of ten records. The time difference between each PSD was 10.24 seconds.

Figure 43 showed how the temporal variations in the PSD are smoothed out by taking successively longer

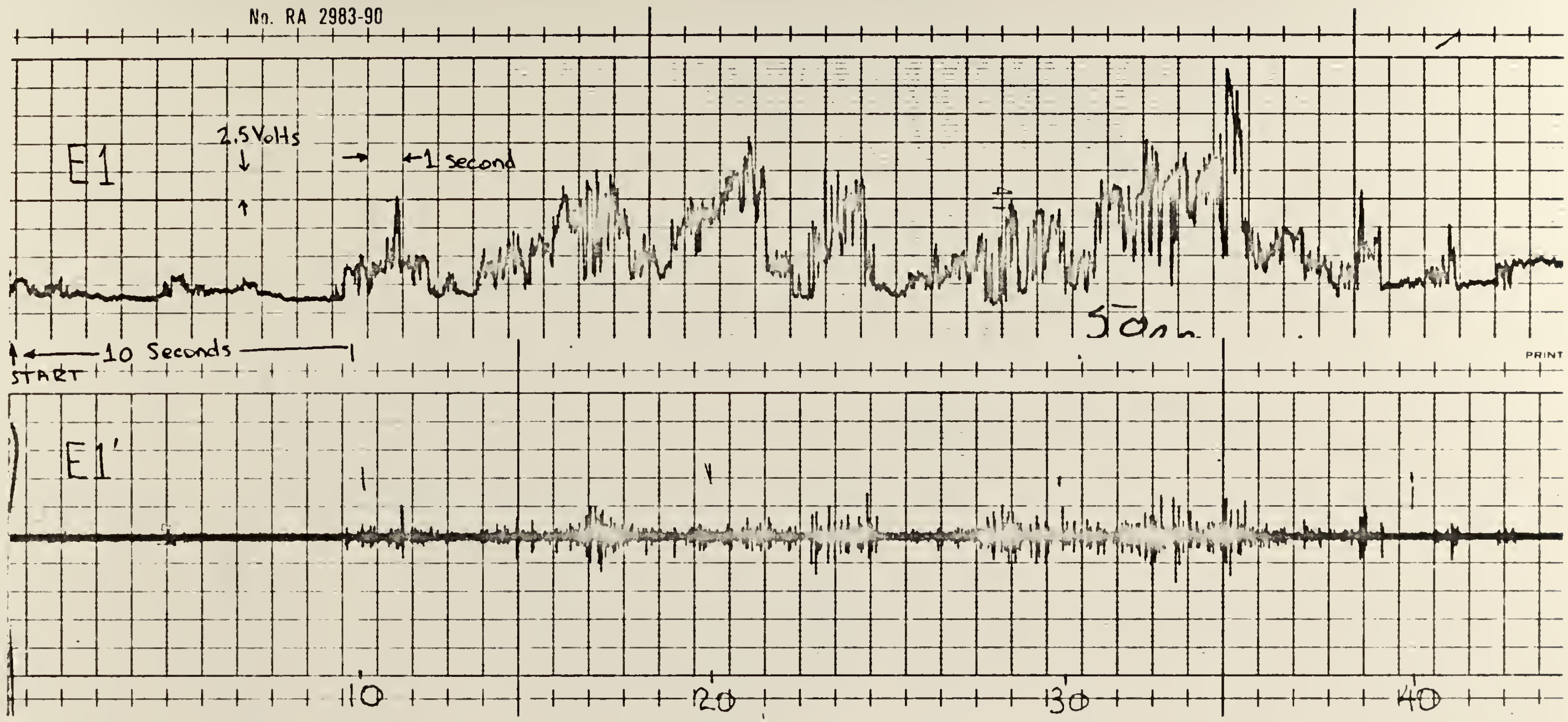


Figure 41. Brush Recording of Undifferentiated and Differentiated Temperature Signal (203(1) E1 and E1')

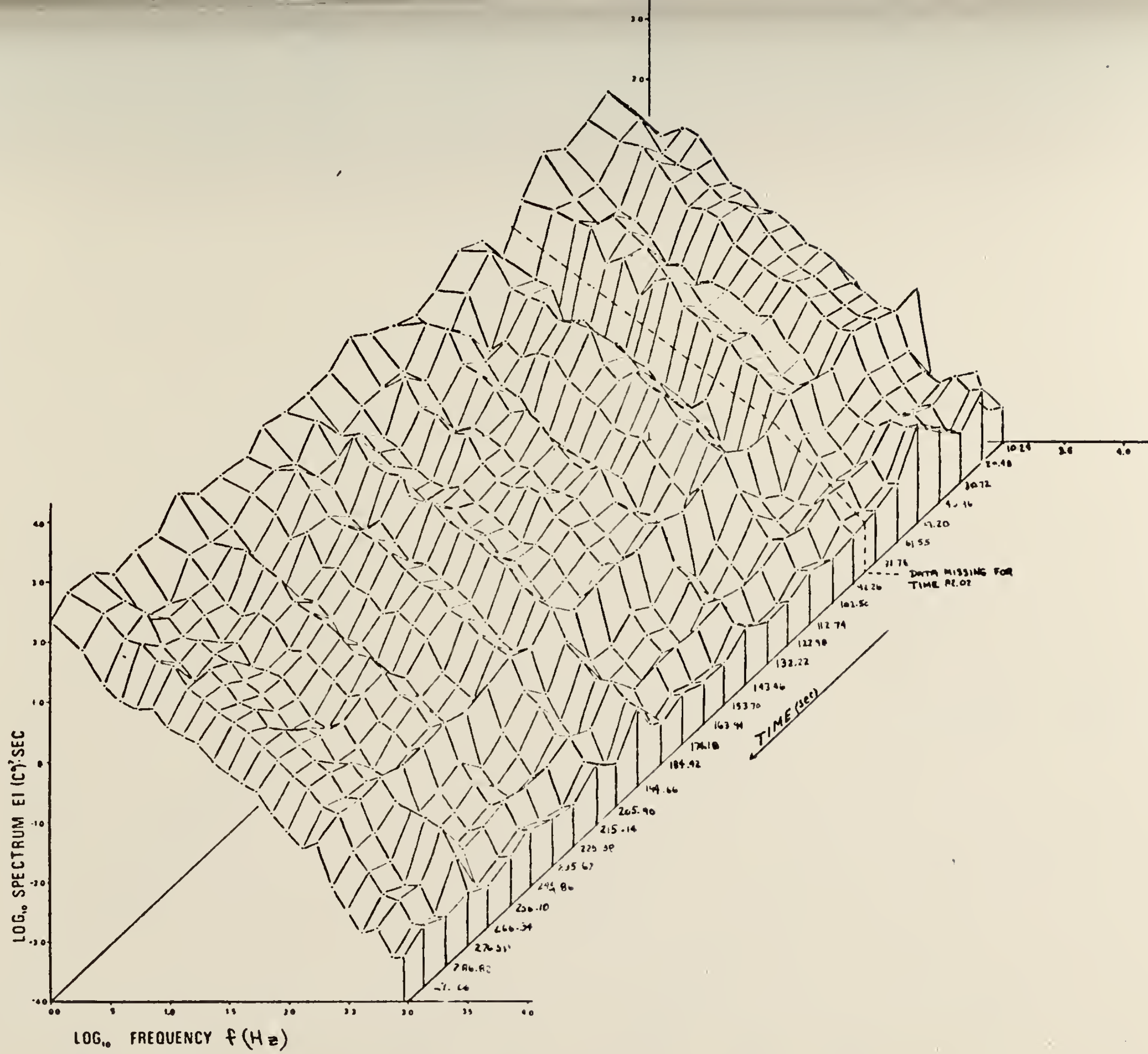


Figure 42. Temporal Variations of Atmospheric Temperature Signal. PSD Computed for Each 10.24 Seconds of Signal

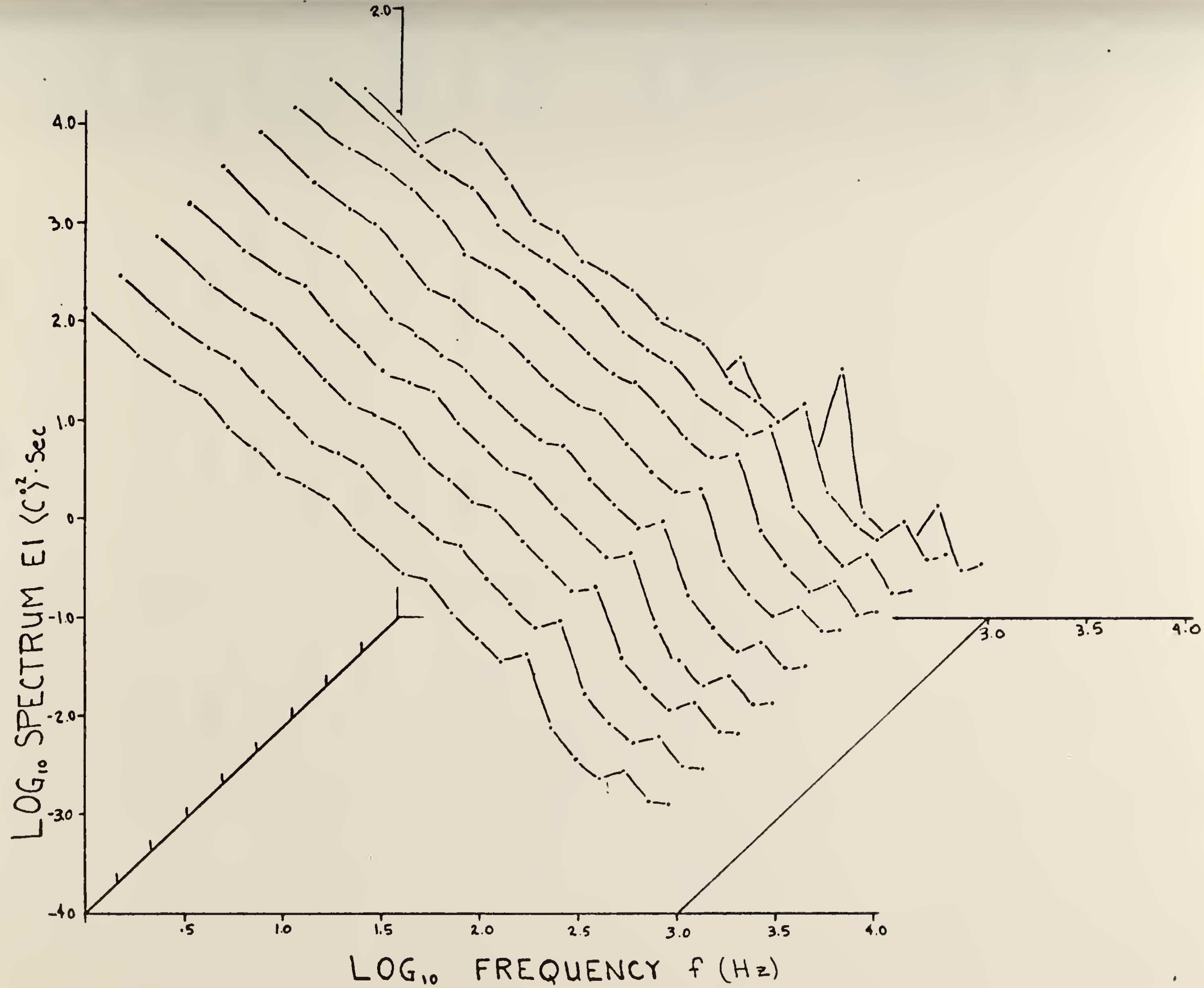


Figure 43. Effect of Increasing Number of Records Analysed for PSD Analysis

records of data. The first PSD was computed from 10 records, starting with record number one. The second PSD along the time axis was computed from 20 records, starting with the first and so forth. The last PSD gave the average of a total of 102.4 seconds of temperature fluctuations, which was computed from 100 records. The smoothing effect was quite evident from the fact that the first PSD was a much lower value, though, its effect was quickly smoothed by the averaging process.

(2) Velocity.

Figure 44 shows the temporal variation of the velocity PSD. A total of 300 records were analyzed. Because of the higher sampling rate of 4000 SPS, the total length of signal shown is only 153.7 seconds:

$$\frac{2048 \text{ Samp/Record}}{4000 \text{ Samp/Sec}} \times 300 \text{ Records}$$

Thirty separate PSD values were computed by sequentially analyzing 10 records at a time. The time difference between each PSD curve was 5.12 seconds.

Figure 45 showed the smoothing effect achieved by analysing an increasing number of records of the velocity signal. The overall trend in smoothing was not as apparent for this signal due to the fact that the original section of signal analyzed gave a good statistical description of the process.

b. PSD Five Minutes of Signal

Figure 46 showed the temperature PSD values for a long length of data (5 minutes) compared with data from a

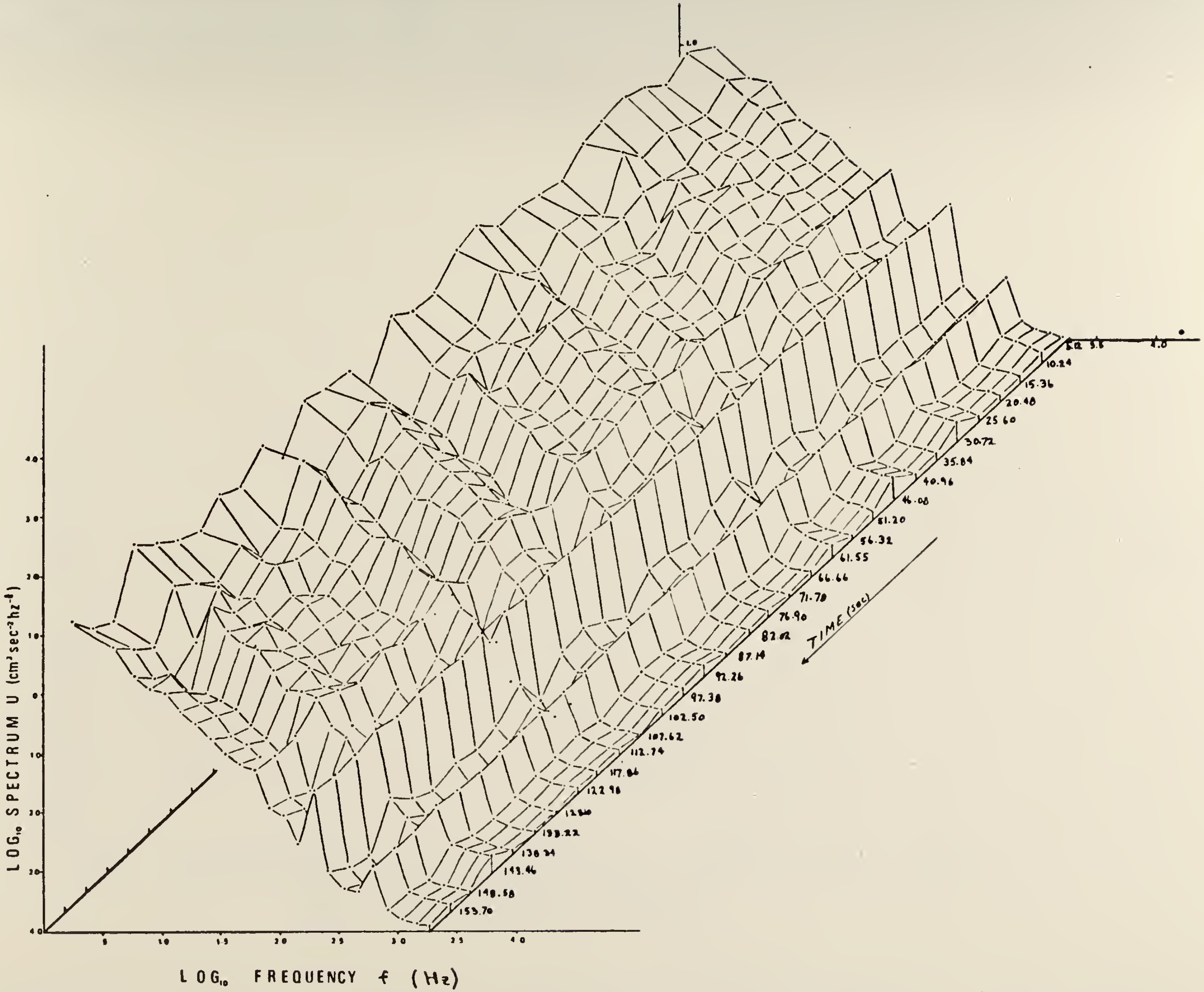


Figure 44. Temporal Variation of Atmospheric Velocity Signal. PSP Computed for Each 5.12 Seconds of Signal

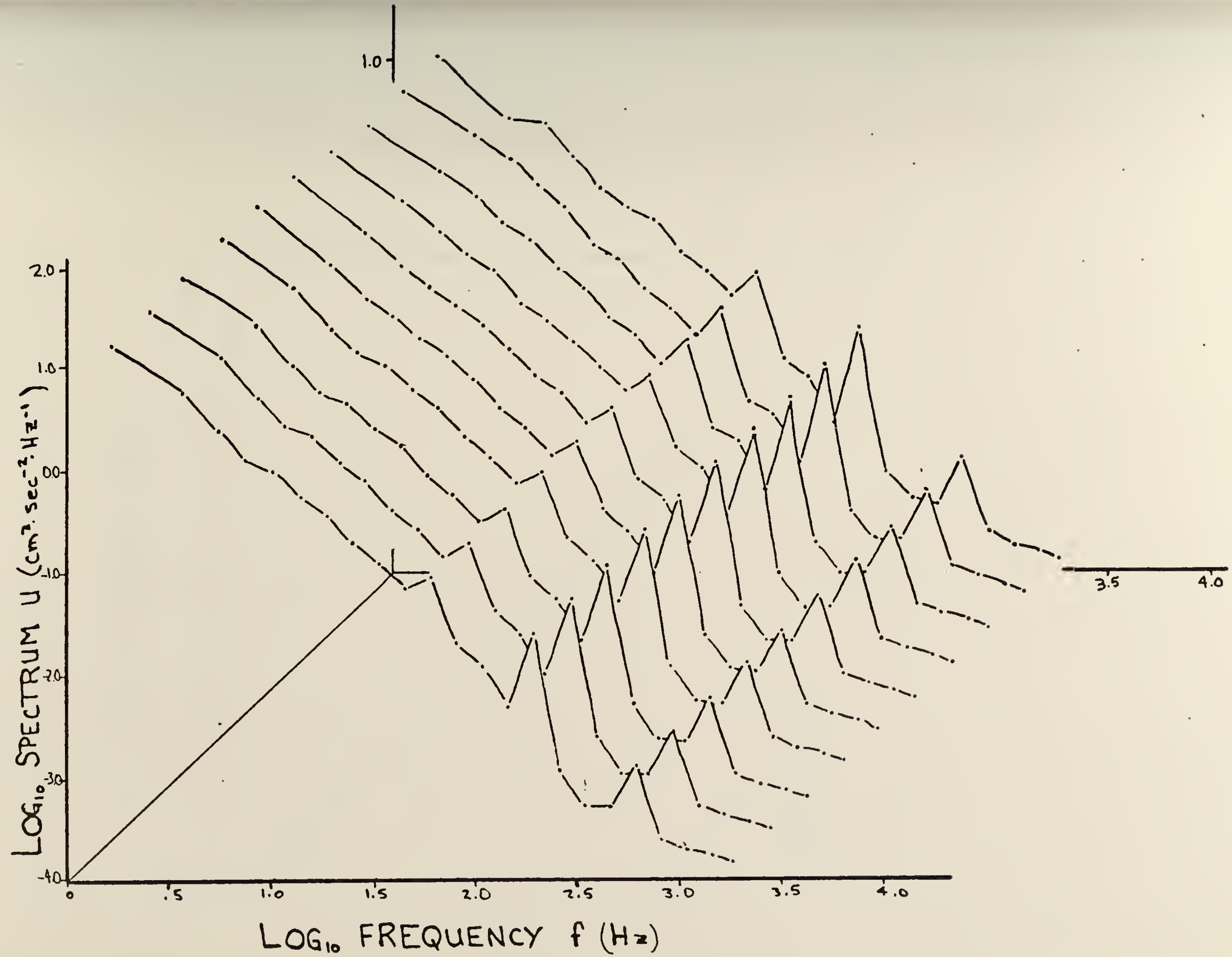


Figure 45. Effect of Increasing Number of Records Analysed for PSD Analysis of Velocity Signal

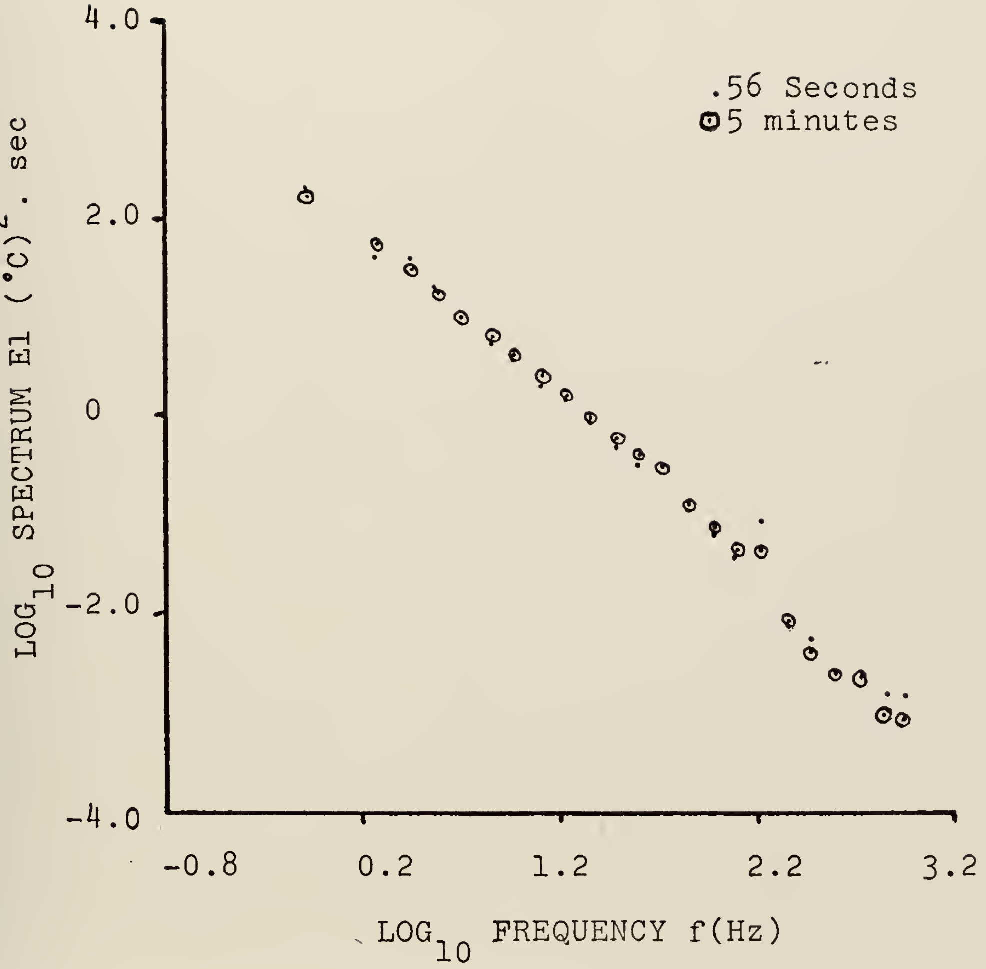


Figure 46. Comparison of 56 Records and 5 Minutes of Temperature Signal

short section of signal. The two lengths compare very closely, implying that for this situation, the shorter length of a minute would have given statistics representative of longer sections, (for the frequency range 1 Hz to 1KHz).

Figure 47 showed the slope characteristics for the slong record length (5 minutes). A definite increase (-7/3) in the slope was noted from about 70 Hz to about 300 Hz.

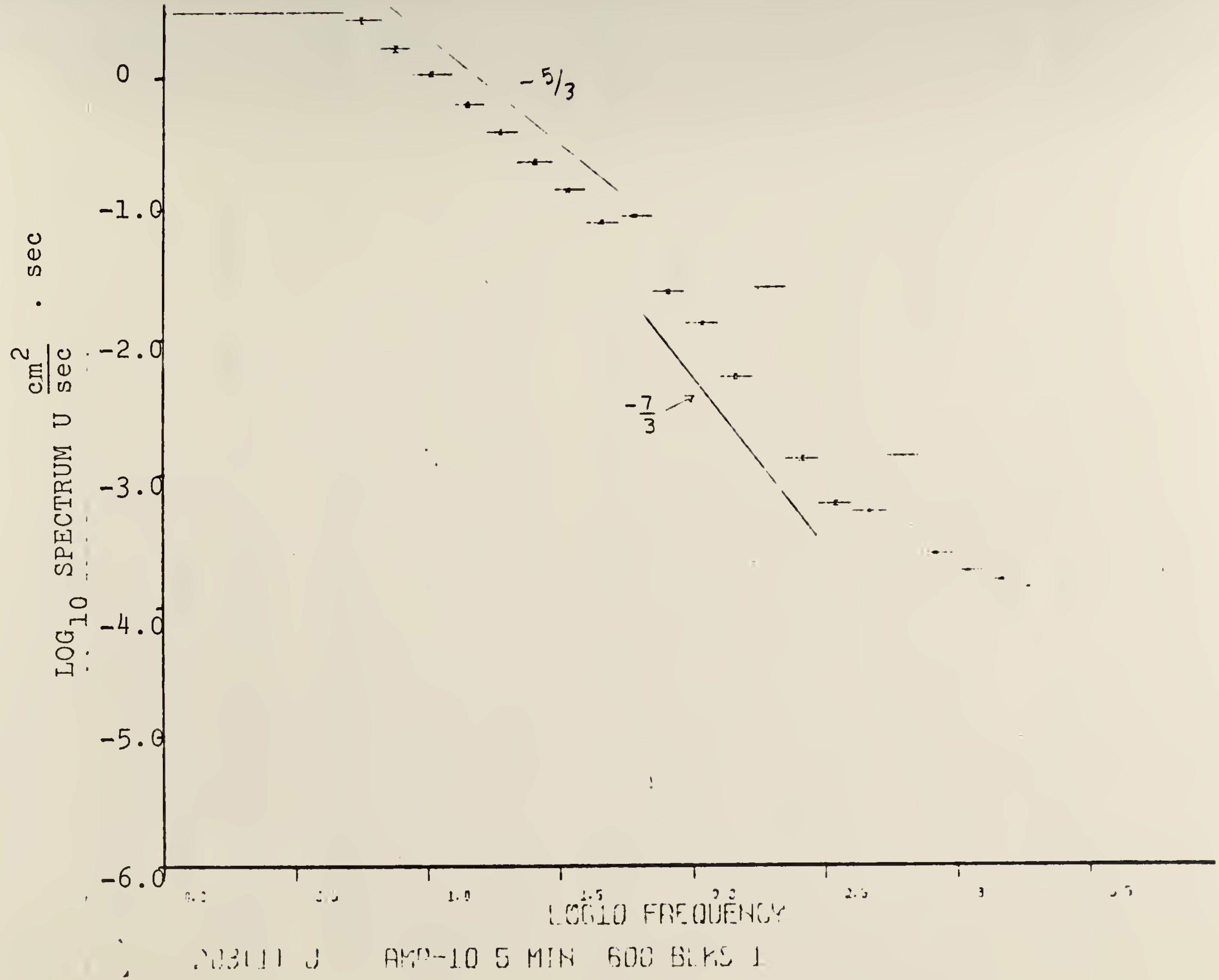


Figure 47. PSD Plot from Analysis of 5 min. of Velocity Signal

VI. CONCLUSIONS AND RECOMMENDATIONS

A. CONCLUSIONS

The following conclusions were reached concerning the problem of the possible input of noise into the Analog-to-Digital conversion procedure and into the PSD analysis procedure with signal inputs of real data.

1. PSD Programs

No significant noise sources were found to exist within the computational program of the Naval Postgraduate School FFT package. The PSD computations for noise -free sine and random signals agreed very well with theory.

The programs FTOR, SCOR and FCPLT were found to be excellent for obtaining power spectra from large quantities of time series data.

In the CONVERT program used in a previous study, it was found that a factor was missing which changed the octal base number to the hexidecimal. A corrected version of the program was used and PSD results indicate the procedure is functioning correctly.

PSD values from four atmospheric turbulence signals which were compared with results obtained from other computational facilities showed very close correlation.

2. Analog-to-Digital Conversion

No significant noise sources were found to be interfering with the digitization procedure carried out on the

Hybrid Computer. Patchboard noise which had previously been reported as excessive by Jones, [Ref. 5] was noted; however, its presence in the PSD plots of sine and random signals were not detected. Even the noise picked up by open 10 gain amplifier had a very low level, on the order of 10^{-6} V^2 Hz^{-1}.

Early in this study it was discovered that the digitization procedure was missing several data samples at the end of each record. The problem was due to internal delays within the XDS 9300 which caused several data samples to be omitted. The computer laboratory staff revised the digitization program to include a cross-check between each sample and lapsed time.

It is now possible to sample at a rate of at least 5000 SPS and check each block for missing data. Results obtained from PSD analysis indicate the problem has been corrected.

3. PSD Analysis Procedures

The IBM 360/67 was found to be quite capable of processing large quantities of time series data. The operational routine in PSD analysis has been improved. Methods for affecting faster "turn around" have been developed.

Several methods have been developed for statistically checking the digital values on tapes and plotting the values by the Calcomp plotter.

4. PSD Analysis of Turbulence Signals

A definite correlation was noted between the temporal PSD plots of both temperature and velocity and the original

analog signal from which the PSD results were computed. This tended to further support the conclusion that the PSD programs were working properly.

B. RECOMMENDATIONS FOR FUTURE WORK

Due to the fact that this series of programs has proved to be an extremely powerful tool for signal analysis, its future use should be vigourously persued. The programs' use in turbulence anlysis has been well established; however, its application in any field which employs PSD techniques should be followed.

The time-varying PSD analysis of turblence signal should be persued. This technique showed interesting possibilities for future work. Digital tapes of four turbulence signals were developed and can be easily assessed for future analysis of the signals. Though a correlation between temperature and velocity was noted, new digital tapes will have to be generated with the dual channel digitization procedure in order to get cross-spectral values for these signals.

```
LEVEL  18                    MAIN                 DATE = 71294
       DIMENSION DATA(2048),Y(50000)
       REWIND 2
       REWIND 4
       IEND=0
       NCHAN=1
       KMAX=2048
       J=0
       A=20.*3.14159
       DO 20 K=1,50000
       T=(K-1.)/4160.
       B=A*T
   20  Y(K)=SIN(B)
   31  J=J+1
       IF(J.LE.1) WRITE(6,70)J
   70  FORMAT('1',10X,'RECORD NO.=',I5)
       IF(J.GT.1)  WRITE(6,71)J
   71  FORMAT(' ',10X,'RECORD NO.=',I5)
       DO 1 K=1,2048
       M=2048*(J-1)+K
    1  DATA(K)=Y(M)
       IF(J.LE.1) WRITE(6,66)(DATA(I),I=1,2048)
   66  FORMAT(1X,8E16.8)
       WRITE(4)KMAX,NCHAN,DATA
       IF(J.EQ.24) GO TO 22
       GO TO 31
   22  END FILE 4
       REWIND 4
       READ(4) KMAX,NCHAN,DATA
       WRITE(6,82)(DATA(I),I=1,2048)
   82  FORMAT(1X,8E16.8)
       STOP
       END
```

APPENDIX A. Program to Generate Digital 10 Hz Sine Wave Samples and Write onto 9-Track Tape

```
/ G LEVEL  18                MAIN                DATE = 71335        01/50/15
          DIMENSION DATA(2048),Y(2048),X(2048),IY(2048)
   C     RANDOM SIGNAL (100BLOCKS) SAMP.RATE=400SPS
          REWIND 2
          REWIND 4
          IEND=0
          NCHAN=1
          KMAX=2048
          IX=3
          J=0
          DO 60 L=1,100
          DO 20 I=1,2048
          CALL RANDU(IX,IP,YFL)
          IX=IP
          X(I)=I
       20 Y(I)=YFL*10**1
       31 J=J+1
          IF(J.LE.1) WRITE(6,70)J
       70 FORMAT('1',10X,'RECORD NO.=',I5)
          IF(J.GT.1)  WRITE(6,71)J
      71  FORMAT(' ',10X,'RECORD NO.=',I5)
          DO 1 N=1,2048
        1 DATA(N)=Y(N)
          IF(J.LE.1) WRITE(6,66)(DATA(I),I=1,2048)
       66 FORMAT(1X,8E16.8)
          WRITE(4)KMAX,NCHAN,DATA
       60 CONTINUE
       22 END FILE 4
          REWIND 4
          READ(4) KMAX,NCHAN,DATA
          WRITE(6,82)(DATA(I),I=1,2048)
      82  FORMAT(1X,8E16.8)
          STOP
          END
```

APPENDIX B. Program to Generate Random Signal and Write onto 9-Track Tape

APPENDIX C

Equipment Specifications

1. Gaussian Noise Generator

Manufactured by	Model No.	Frequency Ranges	Output Spectrum
Elgenco, Inc.	603A	5Hz-20KHz	±1db (10Hz-500KHz)

Output Level	Spectral Density mv/ Hz
20KHz Setting	5 mv/ Hz
3 V_{rms}	@1 V_{rms} output and 20KHz Range

2. Filter

Manufactured by	Model No.	Frequency Range
Khron-Hite, Corp	3340	.001 to 99.9 KHz

Frequency Accuracy	Pass Band Gain	Attenuation Slope
±2%	odb or 20db	-48db/octave

```
SPECTRUM STATISTICS FOR       637 CHANNEL  1  10HZ SINE,S.R.4.16KHZ,10SEC, NO FILT
STATISTICS ARE BASED ON   24 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS   4160.0000SAMP/SEC , MAKING THE BLOCK LENGTH     0.49231 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF   1.000E 00 HAS BEEN APPLIED TO THE INPUT DATA
```

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (V/S	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM (V/S)**2	LAST HARMONIC
1	7.11E 00	8.13E 00	6.09E-02	5.60E-04	-8.890E-06	4.329E-01	5
2	1.52E 01	8.13E 00	4.54E-04	8.73E-05	-1.393E-06	6.921E-03	9
3	2.34E 01	8.13E 00	4.03E-05	2.01E-05	-3.212E-07	9.419E-04	13
4	3.15E 01	8.13E 00	1.61E-05	9.81E-06	-1.564E-07	5.066E-04	17
5	3.96E 01	8.13E 00	8.91E-06	5.90E-06	-9.412E-08	3.529E-04	21
6	4.77E 01	8.13E 00	5.73E-06	3.96E-06	-6.322E-08	2.735E-04	25
7	5.59E 01	8.13E 00	4.02E-06	2.85E-06	-4.553E-08	2.246E-04	29
8	6.40E 01	8.13E 00	2.99E-06	2.15E-06	-3.436E-08	1.912E-04	33
9	7.21E 01	8.13E 00	2.31E-06	1.69E-06	-2.691E-08	1.668E-04	37
10	8.02E 01	8.13E 00	1.85E-06	1.36E-06	-2.164E-08	1.481E-04	41
11	8.84E 01	8.13E 00	1.51E-06	1.12E-06	-1.779E-08	1.333E-04	45
12	9.65E 01	8.13E 00	1.26E-06	9.33E-07	-1.489E-08	1.213E-04	49
13	1.05E 02	8.13E 00	1.06E-06	7.93E-07	-1.264E-08	1.114E-04	53
14	1.13E 02	8.13E 00	9.13E-07	6.82E-07	-1.088E-08	1.029E-04	57
15	1.21E 02	8.13E 00	7.92E-07	5.93E-07	-9.457E-09	9.572E-05	61
16	1.29E 02	8.13E 00	6.94E-07	5.20E-07	-8.296E-09	8.949E-05	65
17	1.37E 02	8.13E 00	6.13E-07	4.60E-07	-7.340E-09	8.402E-05	69
18	1.45E 02	8.13E 00	5.45E-07	4.10E-07	-6.537E-09	7.921E-05	73
19	1.53E 02	8.13E 00	4.89E-07	3.68E-07	-5.861E-09	7.494E-05	77
20	1.61E 02	8.13E 00	4.40E-07	3.32E-07	-5.288E-09	7.110E-05	81
21	1.70E 02	8.13E 00	3.99E-07	3.01E-07	-4.795E-09	6.765E-05	85
22	1.78E 02	8.13E 00	3.63E-07	2.74E-07	-4.368E-09	6.453E-05	89
23	1.86E 02	8.13E 00	3.32E-07	2.50E-07	-3.998E-09	6.168E-05	93
24	1.94E 02	8.13E 00	3.05E-07	2.30E-07	-3.680E-09	5.908E-05	97
25	2.02E 02	8.13E 00	2.81E-07	2.12E-07	-3.379E-09	5.671E-05	101
26	2.10E 02	8.13E 00	2.59E-07	1.96E-07	-3.128E-09	5.452E-05	105
27	2.18E 02	8.13E 00	2.40E-07	1.82E-07	-2.897E-09	5.250E-05	109
28	2.26E 02	8.13E 00	2.24E-07	1.69E-07	-2.702E-09	5.062E-05	113
29	2.35E 02	8.13E 00	2.08E-07	1.58E-07	-2.517E-09	4.888E-05	117
30	2.43E 02	8.13E 00	1.94E-07	1.47E-07	-2.396E-09	4.721E-05	121
31	2.51E 02	8.13E 00	1.82E-07	1.38E-07	-2.189E-09	4.577E-05	125
32	2.59E 02	8.13E 00	1.71E-07	1.30E-07	-2.060E-09	4.435E-05	129

```
INTEGRAL (SUM) UNDER SPECTRUM =   4.99E-01  V/S       **2
ZEROTH HARMONIC (DC) HAS
        AVERAGE =   5.72E-04  V/S
       VARIANCE =   1.30E-04  V/S      **2
          TREND = -5.78E-04  V/S
```

APPENDIX D. PSD Values from Computer Generated 10 Hz Sine Sine Wave

```
SPECTRUM STATISTICS FOR        637 CHANNEL   1   RANDOM SIGNAL +/-10VOLTS 100 BLKS
STATISTICS ARE BASED ON 100 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS  5000.0000SAMP/SEC , MAKING THE BLOCK LENGTH    0.40960 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF  1.000E 00 HAS BEEN APPLIED TO THE INPUT DATA
```

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM ( )**2	LAST HARMONIC
1	7.69E 01	7.81E 01	3.38E-03	5.92E-04	1.849E-06	2.598E-01	47
2	1.55E 02	7.81E 01	3.30E-03	6.41E-04	-1.425E-06	5.109E-01	79
3	2.33E 02	7.81E 01	3.32E-03	6.02E-04	2.848E-06	7.729E-01	111
4	3.11E 02	7.81E 01	3.25E-03	5.66E-04	8.719E-07	1.012E 00	143
5	3.89E 02	7.81E 01	3.37E-03	5.47E-04	8.460E-08	1.312E 00	175
6	4.68E 02	7.81E 01	3.31E-03	5.32E-04	-2.213E-06	1.546E 00	207
7	5.46E 02	7.81E 01	3.35E-03	5.85E-04	1.127E-06	1.826E 00	239
8	6.24E 02	7.81E 01	3.39E-03	5.61E-04	-1.201E-06	2.117E 00	271
9	7.02E 02	7.81E 01	3.34E-03	5.85E-04	-3.397E-06	2.342E 00	303
10	7.80E 02	7.81E 01	3.30E-03	6.37E-04	-1.812E-06	2.577E 00	335
11	8.58E 02	7.81E 01	3.37E-03	5.96E-04	-3.193E-06	2.891E 00	367
12	9.36E 02	7.81E 01	3.32E-03	5.54E-04	1.783E-06	3.108E 00	399
13	1.01E 03	7.81E 01	3.36E-03	6.04E-04	4.943E-07	3.408E 00	431
14	1.09E 03	7.81E 01	3.44E-03	5.58E-04	8.922E-07	3.756E 00	463
15	1.17E 03	7.81E 01	3.35E-03	6.18E-04	-4.148E-07	3.927E 00	495
16	1.25E 03	7.81E 01	3.42E-03	5.94E-04	5.091E-07	4.274E 00	527
17	1.33E 03	7.81E 01	3.31E-03	5.23E-04	-1.410E-07	4.394E 00	559
18	1.41E 03	7.81E 01	3.38E-03	6.58E-04	-2.435E-06	4.754E 00	591
19	1.48E 03	7.81E 01	3.30E-03	5.53E-04	1.446E-06	4.887E 00	623
20	1.56E 03	7.81E 01	3.26E-03	5.31E-04	9.956E-07	5.097E 00	655
21	1.64E 03	7.81E 01	3.33E-03	5.72E-04	-1.279E-06	5.462E 00	687
22	1.72E 03	7.81E 01	3.40E-03	5.62E-04	-5.653E-07	5.841E 00	719
23	1.80E 03	7.81E 01	3.28E-03	5.27E-04	8.046E-07	5.881E 00	751
24	1.87E 03	7.81E 01	3.27E-03	5.72E-04	4.563E-07	6.126E 00	783
25	1.95E 03	7.81E 01	3.28E-03	6.02E-04	-4.385E-06	6.403E 00	815
26	2.03E 03	7.81E 01	3.32E-03	6.56E-04	-1.227E-07	6.749E 00	847
27	2.11E 03	7.81E 01	3.29E-03	5.26E-04	1.501E-06	6.939E 00	879
28	2.19E 03	7.81E 01	3.25E-03	6.08E-04	2.693E-06	7.114E 00	911
29	2.26E 03	7.81E 01	3.45E-03	5.70E-04	1.439E-06	7.816E 00	943
30	2.34E 03	7.81E 01	3.43E-03	5.94E-04	-4.000E-06	8.041E 00	975
31	2.42E 03	7.81E 01	3.32E-03	5.72E-04	3.111E-06	8.036E 00	1007

```
INTEGRAL (SUM) UNDER SPECTRUM =   8.08E 00        **2
ZEROTH HARMONIC (DC) HAD
        AVERAGE =  5.00E 00
       VARIANCE =  3.69E-03          **2
          TREND = -2.97E-04
```

APPENDIX E. PSD values from Computer Generated Random Signal. Sampling Rate = 5000 SPS

SPECTRUM STATISTICS FOR 637 CHANNEL 1 RANDOM SIGNAL +/-10VOLTS 100 BLKS
STATISTICS ARE BASED ON 100 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS 1000.0000SAMP/SEC , MAKING THE BLOCK LENGTH 2.04800 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF 1.000E 00 HAS BEEN APPLIED TO THE INPUT DATA

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM ( )**2	LAST HARMONIC
1	9.52E 00	9.77E 00	1.67E-02	3.04E-03	1.403E-05	1.591E-01	29
2	1.93E 01	9.77E 00	1.68E-02	3.58E-03	8.523E-07	3.246E-01	49
3	2.91E 01	9.77E 00	1.71E-02	3.93E-03	-9.540E-06	4.954E-01	69
4	3.88E 01	9.77E 00	1.63E-02	4.02E-03	1.367E-06	6.331E-01	89
5	4.86E 01	9.77E 00	1.64E-02	3.85E-03	2.185E-05	7.949E-01	109
6	5.83E 01	9.77E 00	1.62E-02	3.50E-03	8.371E-06	9.446E-01	129
7	6.81E 01	9.77E 00	1.64E-02	3.56E-03	3.948E-07	1.120E 00	149
8	7.79E 01	9.77E 00	1.66E-02	3.55E-03	2.756E-06	1.293E 00	169
9	8.76E 01	9.77E 00	1.71E-02	3.98E-03	-6.921E-06	1.503E 00	189
10	9.74E 01	9.77E 00	1.64E-02	3.47E-03	-1.460E-05	1.596E 00	209
11	1.07E 02	9.77E 00	1.65E-02	3.64E-03	-2.690E-07	1.769E 00	229
12	1.17E 02	9.77E 00	1.71E-02	3.00E-03	8.091E-06	2.000E 00	249
13	1.27E 02	9.77E 00	1.71E-02	3.81E-03	-7.200E-06	2.169E 00	269
14	1.36E 02	9.77E 00	1.63E-02	3.35E-03	-1.633E-05	2.219E 00	289
15	1.46E 02	9.77E 00	1.69E-02	4.12E-03	-1.071E-05	2.476E 00	309
16	1.56E 02	9.77E 00	1.67E-02	3.85E-03	-2.525E-05	2.611E 00	329
17	1.66E 02	9.77E 00	1.65E-02	3.58E-03	-1.743E-05	2.738E 00	349
18	1.76E 02	9.77E 00	1.67E-02	3.83E-03	3.417E-06	2.925E 00	369
19	1.85E 02	9.77E 00	1.69E-02	3.65E-03	1.870E-05	3.138E 00	389
20	1.95E 02	9.77E 00	1.66E-02	3.89E-03	3.319E-06	3.237E 00	409
21	2.05E 02	9.77E 00	1.67E-02	4.05E-03	1.658E-06	3.417E 00	429
22	2.15E 02	9.77E 00	1.69E-02	3.71E-03	-7.066E-06	3.636E 00	449
23	2.24E 02	9.77E 00	1.74E-02	3.34E-03	5.557E-06	3.902E 00	469
24	2.34E 02	9.77E 00	1.67E-02	3.84E-03	4.688E-08	3.913E 00	489
25	2.44E 02	9.77E 00	1.72E-02	3.47E-03	2.638E-06	4.188E 00	509
26	2.54E 02	9.77E 00	1.70E-02	3.96E-03	1.429E-06	4.300E 00	529
27	2.63E 02	9.77E 00	1.64E-02	3.65E-03	1.052E-05	4.317E 00	549
28	2.73E 02	9.77E 00	1.65E-02	3.79E-03	-1.914E-05	4.502E 00	569
29	2.83E 02	9.77E 00	1.72E-02	3.76E-03	-7.433E-06	4.862E 00	589
30	2.93E 02	9.77E 00	1.67E-02	3.50E-03	7.813E-06	4.898E 00	609
31	3.02E 02	9.77E 00	1.62E-02	3.44E-03	7.026E-07	4.905E 00	629
32	3.12E 02	9.77E 00	1.66E-02	3.45E-03	2.021E-06	5.182E 00	649

INTEGRAL (SUM) UNDER SPECTRUM = 5.22E 00 **2
ZEROTH HARMONIC (DC) HAD
AVERAGE = 5.00E 00
VARIANCE = 3.69E-03 **2
TREND = -2.97E-0?,

RAN OUT OF HARMONICS

APPENDIX F. PSD Values from Computer Generated Random Signal. Sampling Rate = 1000 SPS

↑
F

SPECTRUM STATISTICS FOR 637 CHANNEL 1 RANDOM SIGNAL 2 VRMS REAL 100 BLKS I
STATISTICS ARE BASED ON 100 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS 5000.0000SAMP/SEC , MAKING THE BLOCK LENGTH 0.40960 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF 1.000E 00 HAS BEEN APPLIED TO THE INPUT DATA

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N)**2/HERTZ	STD.DEV.	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	7.63E 01	7.81E 01	3.29E-06	1.27E-05	1.731E-08	2.533E-04	47
2	1.55E 02	7.81E 01	2.73E-06	1.26E-05	1.766E-08	4.231E-04	79
3	2.33E 02	7.81E 01	2.76E-06	1.27E-05	1.782E-08	6.424E-04	111
4	3.11E 02	7.81E 01	2.74E-06	1.26E-05	1.851E-08	8.534E-04	143
5	3.89E 02	7.81E 01	2.74E-06	1.26E-05	1.815E-08	1.068E-03	175
6	4.68E 02	7.81E 01	2.76E-06	1.27E-05	1.765E-08	1.289E-03	207
7	5.46E 02	7.81E 01	2.76E-06	1.27E-05	1.795E-08	1.504E-03	239
8	6.24E 02	7.81E 01	2.78E-06	1.26E-05	1.879E-08	1.736E-03	271
9	7.02E 02	7.81E 01	2.81E-06	1.28E-05	1.843E-08	1.972E-03	303
10	7.80E 02	7.81E 01	2.81E-06	1.27E-05	1.795E-08	2.190E-03	335
11	8.58E 02	7.81E 01	2.80E-06	1.26E-05	1.730E-08	2.406E-03	367
12	9.36E 02	7.81E 01	2.93E-06	1.26E-05	1.755E-08	2.741E-03	399
13	1.01E 03	7.81E 01	2.68E-05	1.29E-05	1.116E-08	2.722E-02	431
14	1.09E 03	7.81E 01	2.97E-06	1.27E-05	1.731E-08	3.248E-03	463
15	1.17E 03	7.81E 01	2.88E-06	1.26E-05	1.790E-08	3.372E-03	495
16	1.25E 03	7.81E 01	2.90E-06	1.27E-05	1.799E-08	3.622E-03	527
17	1.33E 03	7.81E 01	2.89E-06	1.26E-05	1.689E-08	3.841E-03	559
18	1.41E 03	7.81E 01	2.96E-06	1.27E-05	1.767E-08	4.165E-03	591
19	1.48E 03	7.81E 01	2.94E-06	1.26E-05	1.685E-08	4.366E-03	623
20	1.56E 03	7.81E 01	3.00E-06	1.27E-05	1.814E-08	4.691E-03	655
21	1.64E 03	7.81E 01	2.99E-06	1.26E-05	1.700E-08	4.903E-03	687
22	1.72E 03	7.81E 01	3.00E-06	1.26E-05	1.764E-08	5.147E-03	719
23	1.80E 03	7.81E 01	2.98E-06	1.24E-05	1.932E-08	5.351E-03	751
24	1.87E 03	7.81E 01	2.99E-06	1.26E-05	1.748E-08	5.599E-03	783
25	1.95E 03	7.81E 01	2.97E-06	1.26E-05	1.822E-08	5.790E-03	815
26	2.03E 03	7.81E 01	2.93E-06	1.24E-05	1.777E-08	5.947E-03	847
27	2.11E 03	7.81E 01	2.97E-06	1.26E-05	1.934E-08	6.268E-03	879
28	2.19E 03	7.81E 01	2.92E-06	1.27E-05	1.831E-08	6.386E-03	911
29	2.26E 03	7.81E 01	2.90E-06	1.25E-05	1.833E-08	6.557E-03	943
30	2.34E 03	7.81E 01	2.92E-06	1.28E-05	1.745E-08	6.844E-03	975
31	2.42E 03	7.81E 01	2.89E-06	1.26E-05	1.703E-08	6.985E-03	1007

INTEGRAL (SUM) UNDER SPECTRUM = 8.89E-03 N **2
ZEROTH HARMONIC (DC) HAD
AVERAGE = 1.9 E-02 N
VARIANCE = 1.36E-05 N **2
TREND = 9.45E-06 N
IDIRST= 637 ICMAX= 1 IPMAX=1 IPSTAR= 1 NFILE= 2 IPLOT= 1 IPHASE= 0
LINDCT= 0 STREPEC= 80. BAND= 80. INDOW= 0
ICC KCHA ICH=(1) (2) (3) (4) (5) (6) (7) (8) (9) (10)
1 1 1 0 0 0 0 0 0 0 0 0

APPENDIX G PSD Values for Signal Leakage into Open Amplifier

SPECTRUM STATISTICS FOR 637 CHANNEL 1 RANDOM SIGNAL 2 VRMS REAL 100 BLKS 1
STATISTICS ARE BASED ON 100 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS 5000.0000SAMP/SEC , MAKING THE BLOCK LENGTH 0.40960 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF 1.0000E 00 HAS BEEN APPLIED TO THE INPUT DATA

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N)**2/HERTZ	STD.DEV.	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	7.69E 01	7.81E 01	9.27E-06	3.73E-05	3.258E-08	7.126E-04	47
2	1.55E 02	7.81E 01	8.76E-06	3.71E-05	2.982E-08	1.357E-03	79
3	2.33E 02	7.81E 01	8.82E-06	3.73E-05	3.126E-08	2.056E-03	111
4	3.11E 02	7.81E 01	8.82E-06	3.75E-05	3.093E-08	2.744E-03	143
5	3.89E 02	7.81E 01	8.79E-06	3.72E-05	3.080E-08	3.424E-03	175
6	4.68E 02	7.81E 01	8.78E-06	3.72E-05	3.147E-08	4.104E-03	207
7	5.46E 02	7.81E 01	8.80E-06	3.72E-05	2.971E-08	4.802E-03	239
8	6.24E 02	7.81E 01	8.80E-06	3.72E-05	3.040E-08	5.492E-03	271
9	7.02E 02	7.81E 01	8.85E-06	3.73E-05	3.178E-08	6.209E-03	303
10	7.80E 02	7.81E 01	8.81E-06	3.72E-05	3.206E-08	6.875E-03	335
11	8.58E 02	7.81E 01	8.84E-06	3.70E-05	3.178E-08	7.589E-03	367
12	9.36E 02	7.81E 01	8.92E-06	3.72E-05	3.281E-08	8.355E-03	399
13	1.01E 03	7.81E 01	6.56E-05	4.10E-05	3.518E-08	6.653E-02	431
14	1.09E 03	7.81E 01	9.01E-06	3.72E-05	2.984E-08	9.847E-03	463
15	1.17E 03	7.81E 01	8.88E-06	3.70E-05	3.131E-08	1.040E-02	495
16	1.25E 03	7.81E 01	8.97E-06	3.74E-05	3.115E-08	1.120E-02	527
17	1.33E 03	7.81E 01	8.92E-06	3.72E-05	2.891E-08	1.184E-02	559
18	1.41E 03	7.81E 01	8.98E-06	3.71E-05	3.263E-08	1.262E-02	591
19	1.48E 03	7.81E 01	8.92E-06	3.67E-05	3.111E-08	1.323E-02	623
20	1.56E 03	7.81E 01	9.01E-06	3.72E-05	2.885E-08	1.407E-02	655
21	1.64E 03	7.81E 01	9.01E-06	3.72E-05	2.944E-08	1.477E-02	687
22	1.72E 03	7.81E 01	9.12E-06	3.75E-05	3.402E-08	1.566E-02	719
23	1.80E 03	7.81E 01	8.99E-06	3.71E-05	2.839E-08	1.615E-02	751
24	1.87E 03	7.81E 01	9.04E-06	3.72E-05	3.028E-08	1.694E-02	783
25	1.95E 03	7.81E 01	8.93E-06	3.67E-05	3.059E-08	1.743E-02	815
26	2.03E 03	7.81E 01	8.90E-06	3.68E-05	3.235E-08	1.807E-02	847
27	2.11E 03	7.81E 01	8.97E-06	3.70E-05	3.051E-08	1.891E-02	879
28	2.19E 03	7.81E 01	8.97E-06	3.73E-05	2.829E-08	1.961E-02	911
29	2.26E 03	7.81E 01	9.00E-06	3.76E-05	3.123E-08	2.037E-02	943
30	2.34E 03	7.81E 01	8.95E-06	3.73E-05	2.880E-08	2.096E-02	975
31	2.42E 03	7.81E 01	8.87E-06	3.71E-05	2.995E-08	2.148E-02	1007

INTEGRAL (SUM) UNDER SPECTRUM = 2.61E-02 N **2
ZEROTH HARMONIC (DC) HAD
AVERAGE = 2.34E-02 N
VARIANCE = 1.96E-05 N **2
TREND = 1.49E-05 N
ICTEST= 637 ICMAX= 2 IBMAX=100 IBSTAR= 1 NFILE= 3 IPLOT= 1 IPHASE= 0
LINOCT= 0 STRFREC= 70. BAND= 70. INDOW= 0
ICC KCHA ICH=(1) (2) (3) (4) (5) (6) (7) (8) (9) (10)
1 1 1 2 0 0 0 0 0 0 0 0
2 2 2 1 0 0 0 0 0 0 0 0

APPENDIX H. PSD Values for Signal Leakage into Open Amplifier. Increased Signal Amplitude

```
SPECTRUM STATISTICS FOR      637 CHANNEL  2  RANDOM SIGNAL 2 VRMS REAL 100 BLKS 2
STATISTICS ARE BASED ON  50 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FRECUENCY WAS  4000.0000SAMF/SEC , MAKING THE BLOCK LENGTH     0.51200 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF  1.000E 00 HAS BEEN APPLIED TO THE INPUT DATA
```

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N)**2/HERTZ	STD.DEV.	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	6.84E 01	6.84E 01	2.23E-04	5.06E-05	4.875E-07	1.521E-02	52
2	1.37E 02	6.84E 01	2.05E-04	4.56E-05	6.500E-07	2.800E-02	87
3	2.05E 02	6.84E 01	2.91E-04	5.72E-05	6.211E-07	5.961E-02	122
4	2.73E 02	6.84E 01	1.79E-04	5.62E-05	2.629E-07	4.905E-02	157
5	3.42E 02	6.84E 01	2.46E-04	5.64E-05	3.957E-07	8.398E-02	192
6	4.10E 02	6.84E 01	3.34E-04	6.77E-05	1.103E-06	1.368E-01	227
7	4.79E 02	6.84E 01	2.30E-04	6.21E-05	3.342E-07	1.102E-01	262
8	5.47E 02	6.84E 01	2.57E-04	7.42E-05	4.316E-07	1.404E-01	297
9	6.15E 02	6.84E 01	6.98E-04	1.03E-04	1.280E-06	4.292E-01	332
10	6.84E 02	6.84E 01	2.98E-04	1.17E-04	4.723E-07	2.038E-01	367
11	7.52E 02	6.84E 01	5.88E-04	1.88E-04	1.640E-06	4.419E-01	402
12	8.20E 02	6.84E 01	3.33E-03	3.85E-04	6.847E-06	2.734E 00	437
13	8.89E 02	6.84E 01	1.39E-03	9.23E-04	9.086E-06	1.233E 00	472
14	9.57E 02	6.84E 01	1.29E-02	1.20E-02	1.486E-04	1.234E 01	507
15	1.03E 03	6.84E 01	3.92E 00	2.01E-02	-2.604E-04	4.016E 03	542
16	1.09E 03	6.84E 01	2.32E-03	1.37E-03	9.570E-07	2.539E 00	577
17	1.16E 03	6.84E 01	3.08E-03	3.59E-04	7.416E-06	3.577E 00	612
18	1.23E 03	6.84E 01	6.14E-04	1.81E-04	1.287E-06	7.554E-01	647
19	1.30E 03	6.84E 01	3.07E-04	1.19E-04	9.763E-07	3.990E-01	682
20	1.37E 03	6.84E 01	5.61E-04	1.04E-04	1.086E-06	7.669E-01	717
21	1.44E 03	6.84E 01	4.10E-04	7.10E-05	1.005E-06	5.883E-01	752
22	1.50E 03	6.84E 01	2.07E-04	6.66E-05	4.519E-07	3.108E-01	787
23	1.57E 03	6.84E 01	3.09E-04	4.93E-05	1.541E-07	4.866E-01	822
24	1.64E 03	6.84E 01	2.95E-04	6.52E-05	7.189E-07	4.834E-01	857
25	1.71E 03	6.84E 01	1.85E-04	5.26E-05	6.214E-07	3.161E-01	892
26	1.78E 03	6.84E 01	2.73E-04	5.33E-05	7.708E-07	4.846E-01	927
27	1.85E 03	6.84E 01	2.39E-04	6.52E-05	1.153E-06	4.404E-01	962
28	1.91E 03	6.84E 01	1.74E-04	5.03E-05	4.749E-07	3.328E-01	997

```
INTEGRAL (SUM) UNDER SPECTRUM =  2.70E 02  N       **2
ZEROTH HARMONIC (DC) HAD
      AVERAGE = -3.35E-01 N
     VARIANCE =  1.59E-04 N        **2
        TREND = -5.62E-05 N
ICTEST= 637  ICMAX=  2  IHMAX=100  IHSTAR=  1  NFILE=  4  IPLOT=  1  IPHASE=  0
LINDCT=  0  STRFREQ=        70.   RANG=       70.   INDOW=  0
ICO  KCHN      ICH=(1)   (2)   (3)   (4)   (5)   (6)   (7)   (8)   (9)  (10)
 1    1          1        2     0     0     0     0     0     0     0     0
 2    2          2        1     0     0     0     0     0     0     0     0
```

APPENDIX I. PSD Values for a Real 1000 Hz Sine Wave. Amplitude ±20 Volts.

```
SPECTRUM STATISTICS FOR        637 CHANNEL  2  RANDOM SIGNAL 2 VRMS REAL 100 BLKS 2
STATISTICS ARE BASED ON  52 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS  4000.0000SAMP/SEC , MAKING THE BLOCK LENGTH     0.51200 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF  1.0000 00 HAS BEEN APPLIED TO THE INPUT DATA
```

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	6.84E 01	6.84E 01	5.07E-04	9.76E-05	1.641E-06	3.463E-02	52
2	1.37E 02	6.84E 01	4.76E-04	9.82E-05	9.258E-07	6.514E-02	87
3	2.05E 02	6.84E 01	6.83E-04	1.03E-04	5.900E-07	1.401E-01	122
4	2.73E 02	6.84E 01	3.98E-04	7.19E-05	-7.149E-07	1.089E-01	157
5	3.42E 02	6.84E 01	5.46E-04	9.13E-05	2.681E-07	1.867E-01	192
6	4.10E 02	6.84E 01	7.50E-04	1.38E-04	1.253E-08	3.077E-01	227
7	4.79E 02	6.84E 01	5.07E-04	1.11E-04	-1.166E-06	2.425E-01	262
8	5.47E 02	6.84E 01	5.55E-04	1.09E-04	-1.704E-06	3.037E-01	297
9	6.15E 02	6.84E 01	1.57E-03	1.56E-04	-2.150E-06	9.660E-01	332
10	6.84E 02	6.84E 01	6.06E-04	1.40E-04	-5.445E-07	4.140E-01	367
11	7.52E 02	6.84E 01	1.20E-03	2.50E-04	-2.923E-06	9.056E-01	402
12	8.20E 02	6.84E 01	7.63E-03	6.52E-04	1.779E-06	6.260E 00	437
13	8.89E 02	6.84E 01	2.29E-03	1.26E-03	-2.604E-05	2.031E 00	472
14	9.57E 02	6.84E 01	1.69E-02	1.61E-02	-3.973E-04	1.618E 01	507
15	1.03E 03	6.84E 01	9.29E 00	3.57E-02	1.291E-04	9.523E 03	542
16	1.09E 03	6.84E 01	4.00E-03	1.63E-03	-4.913E-05	4.372E 00	577
17	1.16E 03	6.84E 01	6.94E-03	6.29E-04	-3.171E-06	8.061E 00	612
18	1.23E 03	6.84E 01	1.31E-03	3.02E-04	-6.708E-06	1.617E 00	647
19	1.30E 03	6.84E 01	6.37E-04	1.29E-04	-9.169E-07	8.274E-01	682
20	1.37E 03	6.84E 01	1.26E-03	1.57E-04	-8.881E-07	1.727E 00	717
21	1.44E 03	6.84E 01	9.16E-04	1.37E-04	-1.150E-06	1.315E 00	752
22	1.50E 03	6.84E 01	4.60E-04	1.08E-04	-6.231E-07	6.918E-01	787
23	1.57E 03	6.84E 01	7.16E-04	1.24E-04	-8.463E-07	1.125E 00	822
24	1.64E 03	6.84E 01	6.64E-04	1.02E-04	1.246E-07	1.090E 00	857
25	1.71E 03	6.84E 01	4.17E-04	9.86E-05	-7.461E-07	7.127E-01	892
26	1.78E 03	6.84E 01	6.22E-04	1.10E-04	4.303E-07	1.106E 00	927
27	1.85E 03	6.84E 01	5.51E-04	9.61E-05	-1.522E-07	1.017E 00	962
28	1.91E 03	6.84E 01	3.98E-04	9.27E-05	-2.621E-07	7.621E-01	997

```
INTEGRAL (SUM) UNDER SPECTRUM =  6.39E 02  N      **2
ZEROTH HARMONIC (DC) PSD
        AVERAGE = -6.13E-01 N
       VARIANCE =  4.41E-04 N        **2
          TREND =  3.65E-04 N
```

APPENDIX J. PSD Values for a Real 1000 Hz Sine Wave. Amplitude ±30 Volts.

SPECTRUM STATISTICS FOR 637 CHANNEL 1 RANDOM SIGNAL 5 VRMS REAL 100 BLKS 1
STATISTICS ARE BASED ON 100 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS 5000.0000SAMP/SEC , MAKING THE BLOCK LENGTH 0.40960 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF 1.000E 00 HAS BEEN APPLIED TO THE INPUT DATA

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	7.69E 01	7.81E 01	1.60E-02	2.85E-03	8.532E-06	1.228E 00	47
2	1.55E 02	7.81E 01	1.53E-02	2.86E-03	-5.951E-06	2.379E 00	79
3	2.33E 02	7.81E 01	1.55E-02	2.75E-03	-8.224E-06	3.611E 00	111
4	3.11E 02	7.81E 01	1.54E-02	2.76E-03	-1.608E-05	4.802E 00	143
5	3.89E 02	7.81E 01	1.55E-02	2.61E-03	2.648E-05	6.017E 00	175
6	4.68E 02	7.81E 01	1.54E-02	2.39E-03	4.249E-06	7.218E 00	207
7	5.46E 02	7.81E 01	1.52E-02	2.95E-03	-3.734E-06	8.293E 00	239
8	6.24E 02	7.81E 01	1.54E-02	2.51E-03	-5.992E-06	9.591E 00	271
9	7.02E 02	7.81E 01	1.60E-02	2.80E-03	8.335E-06	1.120E 01	303
10	7.80E 02	7.81E 01	1.55E-02	2.89E-03	-1.258E-05	1.209E 01	335
11	8.58E 02	7.81E 01	1.57E-02	2.90E-03	-7.531E-06	1.346E 01	367
12	9.36E 02	7.81E 01	1.58E-02	2.59E-03	7.839E-06	1.477E 01	399
13	1.01E 03	7.81E 01	1.54E-02	2.68E-03	1.166E-05	1.562E 01	431
14	1.09E 03	7.81E 01	1.65E-02	2.81E-03	6.993E-07	1.798E 01	463
15	1.17E 03	7.81E 01	1.59E-02	2.35E-03	-3.625E-06	1.859E 01	495
16	1.25E 03	7.81E 01	1.58E-02	2.54E-03	5.494E-06	1.975E 01	527
17	1.33E 03	7.81E 01	1.56E-02	2.84E-03	9.764E-06	2.075E 01	559
18	1.41E 03	7.81E 01	1.59E-02	2.81E-03	4.294E-06	2.238E 01	591
19	1.48E 03	7.81E 01	1.53E-02	2.73E-03	4.902E-06	2.271E 01	623
20	1.56E 03	7.81E 01	1.51E-02	2.54E-03	-1.082E-05	2.350E 01	655
21	1.64E 03	7.81E 01	1.37E-02	2.41E-03	1.135E-05	2.251E 01	687
22	1.72E 03	7.81E 01	1.28E-02	2.24E-03	9.061E-06	2.200E 01	719
23	1.80E 03	7.81E 01	1.18E-02	1.91E-03	1.396E-05	2.113E 01	751
24	1.87E 03	7.81E 01	1.01E-02	1.75E-03	1.616E-06	1.886E 01	783
25	1.95E 03	7.81E 01	9.13E-03	1.61E-03	5.864E-06	1.782E 01	815
26	2.03E 03	7.81E 01	7.81E-03	1.37E-03	-1.096E-05	1.586E 01	847
27	2.11E 03	7.81E 01	6.70E-03	1.15E-03	-2.130E-06	1.412E 01	879
28	2.19E 03	7.81E 01	5.95E-03	1.10E-03	-8.555E-06	1.300E 01	911
29	2.26E 03	7.81E 01	5.22E-03	7.92E-04	-2.134E-06	1.182E 01	943
30	2.34E 03	7.81E 01	4.53E-03	7.97E-04	4.124E-07	1.061E 01	975
31	2.42E 03	7.81E 01	4.30E-03	8.29E-04	1.308E-06	1.041E 01	1007

INTEGRAL (SUM) UNDER SPECTRUM = 3.16E 01 N **2
ZEROTH HARMONIC (DC) HAD
AVERAGE = -9.74E-02 N
VARIANCE = 2.55E-03 N **2
TREND = 5.90E-05 N

IDTEST= 637 ICMAX= 2 IBMAX=100 IBSTAR= 1 NFILE= 2 IPLOT= 1 IPHASE= 0
LINOCT= 0 STFFREQ= 7'. BAND= 70. INDOW= 0

ICD	KCHA	ICH=(1)	(2)	(3)	(4)	(5)	(6)	(7)	(8)	(9)	(10)
1	1	1	2	0	0	0	0	0	0	0	0
2	2	2	1	0	0	0	0	0	0	0	0

APPENDIX K. PSD Values for Real Gaussian Signal Elgenco Signal Generator

```
SPECTRUM STATISTICS FOR       637 CHANNEL  2  RANDOM SIGNAL 5 VRMS REAL 100 BLKS 2
STATISTICS ARE BASED ON  50 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS  4000.0000SAMP/SEC , MAKING THE BLOCK LENGTH    0.51200 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF  1.000E 00 HAS BEEN APPLIED TO THE INPUT DATA
```

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	6.84E 01	6.84E 01	9.20E-02	1.58E-02	-4.112E-05	6.291E 00	52
2	1.37E 02	6.84E 01	4.76E-02	8.51E-03	-4.881E-05	6.508E 00	87
3	2.05E 02	6.84E 01	3.04E-02	5.52E-03	5.712E-05	6.231E 00	122
4	2.73E 02	6.84E 01	2.36E-02	3.46E-03	-7.304E-06	6.462E 00	157
5	3.42E 02	6.84E 01	1.84E-02	3.12E-03	-2.774E-05	6.292E 00	192
6	4.10E 02	6.84E 01	1.61E-02	2.41E-03	-5.512E-05	6.591E 00	227
7	4.79E 02	6.84E 01	1.37E-02	2.63E-03	-9.043E-06	6.550E 00	262
8	5.47E 02	6.84E 01	1.26E-02	1.91E-03	5.069E-06	6.879E 00	297
9	6.15E 02	6.84E 01	1.09E-02	1.51E-03	2.286E-05	6.688E 00	332
10	6.84E 02	6.84E 01	1.03E-02	1.73E-03	-3.367E-05	7.046E 00	367
11	7.52E 02	6.84E 01	9.39E-03	1.45E-03	1.423E-05	7.060E 00	402
12	8.20E 02	6.84E 01	8.77E-03	1.66E-03	2.049E-05	7.194E 00	437
13	8.89E 02	6.84E 01	7.74E-03	1.15E-03	-8.356E-06	6.877E 00	472
14	9.57E 02	6.84E 01	7.43E-03	9.81E-04	1.024E-06	7.109E 00	507
15	1.03E 03	6.84E 01	7.06E-03	1.16E-03	2.472E-06	7.239E 00	542
16	1.09E 03	6.84E 01	6.63E-03	1.24E-03	3.474E-06	7.247E 00	577
17	1.16E 03	6.84E 01	6.33E-03	1.07E-03	8.787E-06	7.361E 00	612
18	1.23E 03	6.84E 01	5.95E-03	1.16E-03	-3.274E-06	7.315E 00	647
19	1.30E 03	6.84E 01	5.41E-03	8.47E-04	7.673E-07	7.022E 00	682
20	1.37E 03	6.84E 01	5.26E-03	7.12E-04	3.572E-09	7.192E 00	717
21	1.44E 03	6.84E 01	4.82E-03	7.79E-04	1.283E-05	6.925E 00	752
22	1.50E 03	6.84E 01	4.49E-03	9.46E-04	-1.430E-05	6.750E 00	787
23	1.57E 03	6.84E 01	4.11E-03	6.33E-04	-4.686E-06	6.469E 00	822
24	1.64E 03	6.84E 01	3.77E-03	5.54E-04	4.591E-06	6.190E 00	857
25	1.71E 03	6.84E 01	3.45E-03	5.65E-04	-1.091E-06	5.892E 00	892
26	1.78E 03	6.84E 01	3.13E-03	5.17E-04	-3.760E-06	5.568E 00	927
27	1.85E 03	6.84E 01	2.92E-03	4.87E-04	-3.146E-06	5.396E 00	962
28	1.91E 03	6.84E 01	2.89E-03	4.67E-04	-3.797E-06	5.526E 00	997

```
INTEGRAL (SUM) UNDER SPECTRUM =   2.56E 01   N          **2
ZEROTH HARMONIC (DC) HAS
        AVERAGE =   9.83E-01  N
       VARIANCE =   7.45E-02  N        **2
          TREND = -1.53E-03  N
```

APPENDIX L PSD Values for Real Gaussian Signal: Ci 5000 Random Signal Generator (HF)

SPECTRUM STATISTICS FOR 637 CHANNEL 1 203(1) E1 A AMP=10 56 BLKS 1
STATISTICS ARE BASED ON 56 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS 2000.0000SAMP/SEC , MAKING THE BLOCK LENGTH 1.02400 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF 1.000E 01 HAS BEEN APPLIED TO THE INPUT DATA

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N)**2/HERTZ	STD.DEV.	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	8.46E-01	9.77E-01	2.01E 02	4.39E 02	5.368E 00	1.701E 02	1
2	1.89E 00	9.77E-01	4.35E 01	7.38E 01	6.153E-01	8.229E 01	2
3	2.89E 00	9.77E-01	4.05E 01	8.10E 01	5.389E-01	1.170E 02	3
4	3.88E 00	9.77E-01	2.13E 01	4.53E 01	1.498E-01	8.267E 01	4
5	5.28E 00	1.95E 00	1.08E 01	1.69E 01	9.121E-02	5.686E 01	6
6	7.26E 00	1.95E 00	5.58E 00	8.27E 00	1.049E-01	4.052E 01	8
7	9.66E 00	2.93E 00	3.77E 00	6.18E 00	2.878E-02	3.638E 01	11
8	1.30E 01	3.91E 00	1.97E 00	2.74E 00	2.715E-02	2.562E 01	15
9	1.74E 01	4.89E 00	1.62E 00	2.30E 00	2.482E-02	2.817E 01	20
10	2.32E 01	6.84E 00	8.36E-01	1.02E 00	9.195E-03	1.939E 01	27
11	3.09E 01	8.79E 00	4.60E-01	5.87E-01	7.794E-03	1.422E 01	36
12	4.11E 01	1.17E 01	3.01E-01	4.25E-01	3.796E-03	1.237E 01	48
13	5.46E 01	1.56E 01	3.00E-01	3.02E-01	2.086E-03	1.637E 01	64
14	7.29E 01	2.15E 01	1.18E-01	1.74E-01	2.039E-03	8.633E 00	86
15	9.76E 01	2.83E 01	6.39E-02	8.88E-02	7.489E-04	6.234E 00	115
16	1.30E 02	3.71E 01	3.87E-02	5.27E-02	4.247E-04	5.032E 00	153
17	1.73E 02	4.98E 01	8.14E-02	2.88E-02	2.941E-04	1.409E 01	204
18	2.31E 02	6.74E 01	7.55E-03	9.81E-03	1.071E-04	1.744E 00	273
19	3.08E 02	8.89E 01	5.19E-03	5.26E-03	5.209E-05	1.600E 00	364
20	4.11E 02	1.19E 02	2.40E-03	4.15E-03	3.859E-05	9.849E-01	486
21	5.49E 02	1.58E 02	2.21E-03	3.51E-03	2.808E-05	1.210E 00	648
22	7.31E 02	2.11E 02	1.53E-03	4.03E-03	3.110E-05	1.115E 00	864
23	9.19E 02	1.56E 02	1.45E-03	4.14E-03	3.007E-05	1.332E 00	1024

INTEGRAL (SUM) UNDER SPECTRUM = 3.88E 02 N **2
ZEROTH HARMONIC (DC) HAD
AVERAGE = -5.09E 01 N
VARIANCE = 9.06E 02 N **2
TREND = 5.45E-01 N

APPENDIX M. PSD Values for Temperature Signal: 57.12 Seconds of 203(1) E1(A)

SPECTRUM STATISTICS FOR 637 CHANNEL 1 203(1) E1' AMP=10 5 MIN 600 BLKS 1
STATISTICS ARE BASED ON 600 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS 4000.000 SAMP/SEC , MAKING THE BLOCK LENGTH 0.51200 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF 1.000E 01 HAS BEEN APPLIED TO THE INPUT DATA

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	1.69E 00	1.95E 00	1.47E-02	2.91E-02	1.527E-05	2.493E-02	1
2	3.78E 00	1.95E 00	2.07E-02	4.47E-02	6.617E-07	7.841E-02	2
3	5.79E 00	1.95E 00	2.12E-02	4.56E-02	1.890E-05	1.226E-01	3
4	7.75E 00	1.95E 00	2.27E-02	5.29E-02	8.328E-06	1.756E-01	4
5	1.06E 01	3.91E 00	2.94E-02	7.13E-02	2.528E-05	3.102E-01	6
6	1.45E 01	3.91E 00	3.49E-02	7.14E-02	1.093E-05	5.072E-01	8
7	1.93E 01	5.86E 00	3.78E-02	7.46E-02	3.385E-05	7.305E-01	11
8	2.61E 01	7.81E 00	3.99E-02	6.49E-02	2.797E-05	1.040E 00	15
9	3.48E 01	9.77E 00	4.49E-02	8.05E-02	4.963E-05	1.562E 00	20
10	4.64E 01	1.37E 01	5.24E-02	9.72E-02	4.958E-05	2.430E 00	27
11	6.19E 01	1.76E 01	1.26E-01	8.79E-02	8.856E-05	7.789E 00	36
12	8.22E 01	2.34E 01	5.44E-02	9.41E-02	4.673E-05	4.472E 00	48
13	1.09E 02	3.13E 01	4.91E-02	8.66E-02	2.634E-05	5.363E 00	64
14	1.46E 02	4.30E 01	4.38E-02	8.87E-02	2.686E-05	6.392E 00	86
15	1.95E 02	5.66E 01	1.34E-01	5.80E-02	2.709E-06	2.622E 01	115
16	2.60E 02	7.42E 01	2.15E-02	3.82E-02	2.045E-05	5.587E 00	153
17	3.46E 02	9.96E 01	1.31E-02	2.20E-02	1.078E-05	4.524E 00	204
18	4.62E 02	1.35E 02	7.11E-03	1.07E-02	3.446E-06	3.284E 00	273
19	6.17E 02	1.78E 02	6.99E-03	5.91E-03	7.697E-07	4.311E 00	364
20	8.22E 02	2.33E 02	5.87E-03	3.11E-03	-3.498E-08	4.831E 00	486
21	1.10E 03	3.16E 02	6.35E-03	1.56E-03	-5.011E-07	6.968E 00	649
22	1.46E 03	4.22E 02	6.18E-03	1.05E-03	-4.172E-07	9.039E 00	864
23	1.84E 03	3.13E 02	4.94E-03	6.59E-04	-2.544E-07	9.084E 00	1024

INTEGRAL (SUM) UNDER SPECTRUM = 2.93E 01 N **2
ZEROTH HARMONIC (DC) HAD
AVERAGE = 2.44E 00 N
VARIANCE = 2.53E-02 N **2
TREND = 6.45E-04 N

APPENDIX N. PSD Values for NPS Analysis of Differentiated Temperature Signal:5 minutes of 203(1) E1'

```
SPECTRUM STATISTICS FOR 10   637 CHANNEL  1  203(1) U    AMP=10 5 MIN 10 BLKS 1
STATISTICS ARE BASED ON 10 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS  4000.0000SAMP/SEC , MAKING THE BLOCK LENGTH    0.51200 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF  1.000E 01 HAS BEEN APPLIED TO THE INPUT DATA
```

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	1.69E 00	1.95E 00	1.99E 01	2.96E 01	1.564E-02	3.358E 01	1
2	3.78E 00	1.95E 00	6.21E 00	9.68E 00	4.754E-03	2.347E 01	2
3	5.78E 00	1.95E 00	2.73E 00	3.96E 00	2.095E-03	1.578E 01	3
4	7.75E 00	1.95E 00	1.65E 00	2.15E 00	1.244E-03	1.277E 01	4
5	1.06E 01	3.91E 00	1.05E 00	1.21E 00	5.901E-04	1.109E 01	6
6	1.45E 01	3.91E 00	6.15E-01	6.75E-01	5.807E-04	8.924E 00	8
7	1.93E 01	5.86E 00	3.82E-01	3.86E-01	2.853E-04	7.373E 00	11
8	2.61E 01	7.81E 00	2.25E-01	2.36E-01	2.203E-04	5.871E 00	15
9	3.48E 01	9.77E 00	1.37E-01	1.34E-01	1.434E-04	4.763E 00	20
10	4.64E 01	1.37E 01	7.84E-02	7.40E-02	6.730E-05	3.637E 00	27
11	6.19E 01	1.76E 01	8.71E-02	5.10E-02	5.037E-05	5.390E 00	36
12	8.22E 01	2.34E 01	2.32E-02	2.47E-02	2.798E-05	1.911E 00	48
13	1.09E 02	3.13E 01	1.34E-02	1.29E-02	1.240E-05	1.462E 00	64
14	1.46E 02	4.30E 01	5.20E-03	5.95E-03	6.351E-06	7.590E-01	86
15	1.95E 02	5.66E 01	2.50E-02	3.86E-03	2.040E-08	4.890E 00	115
16	2.60E 02	7.42E 01	1.26E-03	1.28E-03	1.034E-06	3.285E-01	153
17	3.46E 02	9.96E 01	5.78E-04	5.77E-04	4.173E-07	2.000E-01	204
18	4.62E 02	1.35E 02	4.99E-04	3.15E-04	1.391E-07	2.303E-01	273
19	6.17E 02	1.78E 02	1.31E-03	1.98E-04	1.151E-07	8.103E-01	364
20	8.22E 02	2.38E 02	2.40E-04	1.15E-04	4.797E-08	1.976E-01	486
21	1.10E 03	3.16E 02	1.76E-04	8.47E-05	1.030E-08	1.933E-01	648
22	1.46E 03	4.22E 02	1.50E-04	6.92E-05	8.045E-10	2.198E-01	864
23	1.84E 03	3.13E 02	1.33E-04	6.54E-05	-1.331E-10	2.445E-01	1024

```
INTEGRAL (SUM) UNDER SPECTRUM =  7.72E 01  N       **2
ZEROTH HARMONIC (DC) HAD
        AVERAGE =   1.05E 01 N
       VARIANCE =   4.28E 02 N       **2
          TREND =   4.71E-03 N
IDTEST= 637  ICMAX=  1  IBMAX=600  IBSTAR=  1  NFILF=  2  IPLOT= 13  IPHASE=  0
LINOCT=  1  STRFREQ=      0.   BAND=       0.   INDOW=  0.
ICD  KCHA    ICH=(1)   (2)   (3)   (4)   (5)   (6)   (7)   (8)   (9)   (10)
 1     1        1       0     0     0     0     0     0     0     0     0
```

APPENDIX O. PSD Values for NPS Analysis of Velocity Signal: 51.2 seconds of 203(1) U(A)

```
SPECTRUM STATISTICS FOR       637 CHANNEL  1  203(1) U'    AMP=10 5 MIN  600 BLKS 1
STATISTICS ARE BASED ON 600 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS  4000.0000SAMP/SEC . MAKING THE BLOCK LENGTH    0.51200 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF  1.000E 01 HAS BEEN APPLIED TO THE INPUT DATA
```

	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	1.69E 00	1.95E 00	6.18E-02	8.55E-02	3.467E-05	1.045E-01	1
2	3.78E 00	1.95E 00	7.86E-02	1.28E-01	3.533E-05	2.974E-01	2
3	5.78E 00	1.95E 00	8.85E-02	1.21E-01	3.135E-05	5.113E-01	3
4	7.75E 00	1.95E 00	9.17E-02	1.38E-01	1.330E-04	7.105E-01	4
5	1.06E 01	3.91E 00	1.10E-01	1.26E-01	4.700E-05	1.161E 00	6
6	1.45E 01	3.91E 00	1.21E-01	1.28E-01	1.051E-04	1.756E 00	8
7	1.93E 01	5.86E 00	1.39E-01	1.34E-01	9.931E-05	2.685E 00	11
8	2.61E 01	7.81E 00	1.55E-01	1.59E-01	1.360E-04	4.048E 00	15
9	3.48E 01	9.77E 00	1.68E-01	1.69E-01	1.761E-04	5.851E 00	20
10	4.64E 01	1.37E 01	1.69E-01	1.52E-01	1.339E-04	7.853E 00	27
11	6.19E 01	1.76E 01	2.08E-01	1.81E-01	1.922E-04	1.288E 01	36
12	8.22E 01	2.34E 01	1.53E-01	1.77E-01	2.096E-04	1.257E 01	48
13	1.09E 02	3.13E 01	1.23E-01	1.49E-01	1.661E-04	1.342E 01	64
14	1.46E 02	4.30E 01	9.11E-02	1.20E-01	1.213E-04	1.329E 01	86
15	1.95E 02	5.66E 01	7.46E-02	8.00E-02	8.898E-05	1.457E 01	115
16	2.60E 02	7.42E 01	3.27E-02	6.63E-02	5.309E-05	8.497E 00	153
17	3.46E 02	9.96E 01	1.53E-02	2.79E-02	2.621E-05	5.287E 00	204
18	4.62E 02	1.35E 02	6.55E-03	1.73E-02	1.097E-05	3.026E 00	273
19	6.17E 02	1.78E 02	2.89E-03	5.21E-03	3.590E-06	1.782E 00	364
20	8.22E 02	2.38E 02	7.54E-04	2.19E-03	8.181E-07	6.203E-01	486
21	1.10E 03	3.16E 02	4.36E-04	1.34E-03	1.138E-07	4.788E-01	648
22	1.46E 03	4.22E 02	5.23E-04	9.92E-04	-5.092E-08	7.648E-01	864
23	1.84E 03	3.13E 02	6.69E-04	8.50E-04	-4.049E-08	1.230E 00	1024

```
INTEGRAL (SUM) UNDER SPECTRUM =   3.28E 01  N        **2
ZEROTH HARMONIC (DC) HAD
        AVERAGE =  6.65E 00 N
        VARIANCE =  6.29E-02 N       **2
        TREND =  8.55E-04 N
```

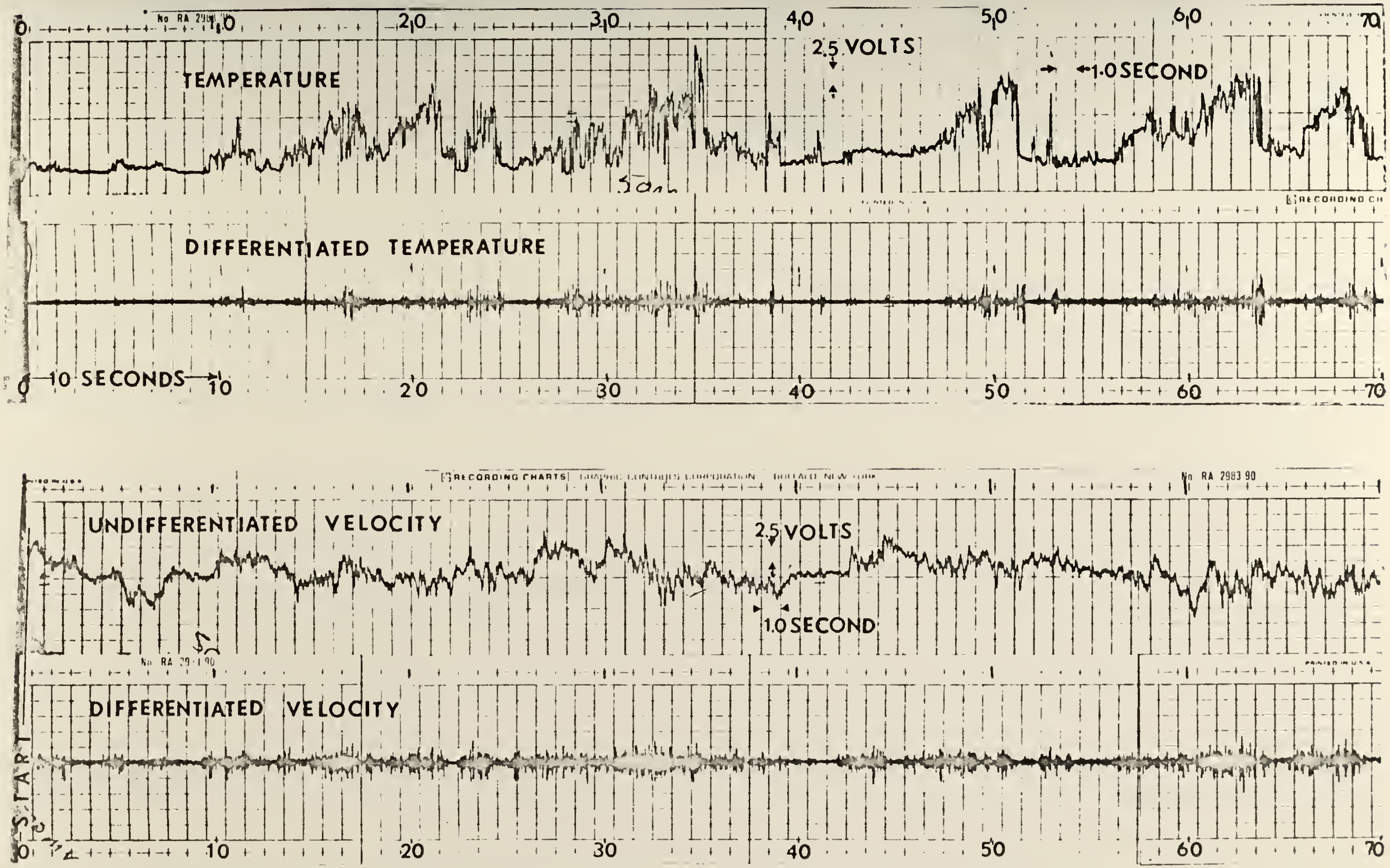
APPENDIX P. PSD Values for NPS Analysis of Differentiated Velocity Signal: 5 Minutes of 203(1) U

SPECTRUM STATISTICS FOR 637 CHANNEL 1 203(1) EI A AMP=10 300 BLKS 1
STATISTICS ARE BASED ON 300 BLOCKS OF 2048 SAMPLES EACH
THE SAMPLING FREQUENCY WAS 2000.0000SAMP/SEC , MAKING THE BLOCK LENGTH 1.02400 SECONDS
TREND IS THE AVERAGE OF (VALUE(A)-VALUE(B))/(BLOCK NO.(A)-BLOCK NO.(B))
A CALIBRATION FACTOR OF 1.000E 01 HAS BEEN APPLIED TO THE INPUT DATA

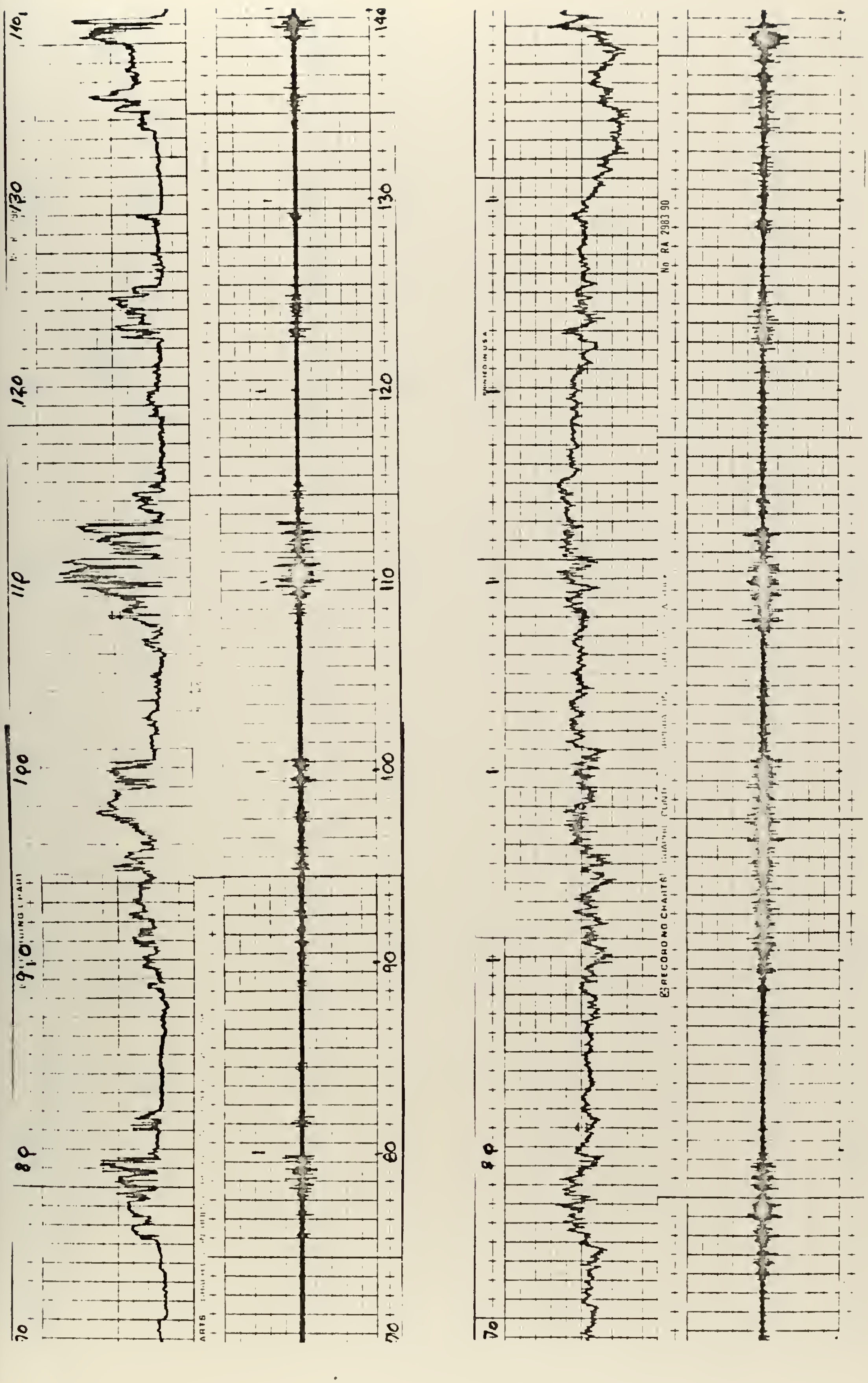
	FREQUENCY HERTZ	BANDWIDTH	SPECTRUM (N	STD.DEV.)**2/HERTZ	TREND	FREQ*SPECTRUM (N)**2	LAST HARMONIC
1	9.46E-01	9.77E-01	1.82E 02	3.62E 02	5.466E-01	1.537E 02	1
2	1.99E 00	9.77E-01	5.76E 01	1.13E 02	8.521E-02	1.089E 02	2
3	2.89E 00	9.77E-01	3.13E 01	5.95E 01	4.054E-02	9.034E 01	3
4	3.88E 00	9.77E-01	1.90E 01	3.69E 01	6.466E-03	7.360E 01	4
5	5.28E 00	1.95E 00	9.95E 00	1.53E 01	9.225E-03	5.253E 01	6
6	7.26E 00	1.95E 00	6.23E 00	9.43E 00	1.318E-02	4.523E 01	8
7	9.66E 00	2.93E 00	4.23E 00	6.81E 00	9.243E-03	4.080E 01	11
8	1.30E 01	3.91E 00	2.63E 00	4.51E 00	2.747E-03	3.425E 01	15
9	1.74E 01	4.88E 00	1.68E 00	3.00E 00	1.885E-03	2.924E 01	20
10	2.32E 01	6.84E 00	9.45E-01	1.23E 00	1.936E-03	2.190E 01	27
11	3.09E 01	8.79E 00	6.06E-01	9.26E-01	9.232E-04	1.876E 01	36
12	4.11E 01	1.17E 01	3.99E-01	6.23E-01	9.193E-04	1.638E 01	48
13	5.46E 01	1.56E 01	2.83E-01	3.50E-01	3.550E-04	1.544E 01	64
14	7.29E 01	2.15E 01	1.34E-01	1.90E-01	1.935E-04	9.806E 00	86
15	9.76E 01	2.83E 01	7.61E-02	1.12E-01	1.344E-04	7.428E 00	115
16	1.30E 02	3.71E 01	4.13E-02	6.17E-02	4.499E-05	5.376E 00	153
17	1.73E 02	4.98E 01	4.51E-02	2.88E-02	3.031E-05	7.796E 00	204
18	2.31E 02	6.74E 01	8.76E-03	1.41E-02	1.711E-05	2.022E 00	273
19	3.09E 02	8.89E 01	4.15E-03	9.32E-03	8.176E-06	1.281E 00	364
20	4.11E 02	1.19E 02	2.24E-03	7.08E-03	3.409E-06	9.222E-01	486
21	5.49E 02	1.58E 02	2.50E-03	4.84E-03	-1.583E-07	1.370E 00	648
22	7.31E 02	2.11E 02	9.53E-04	2.84E-03	-1.951E-06	6.969E-01	864
23	9.19E 02	1.56E 02	8.43E-04	2.21E-03	-2.764E-06	7.749E-01	1024

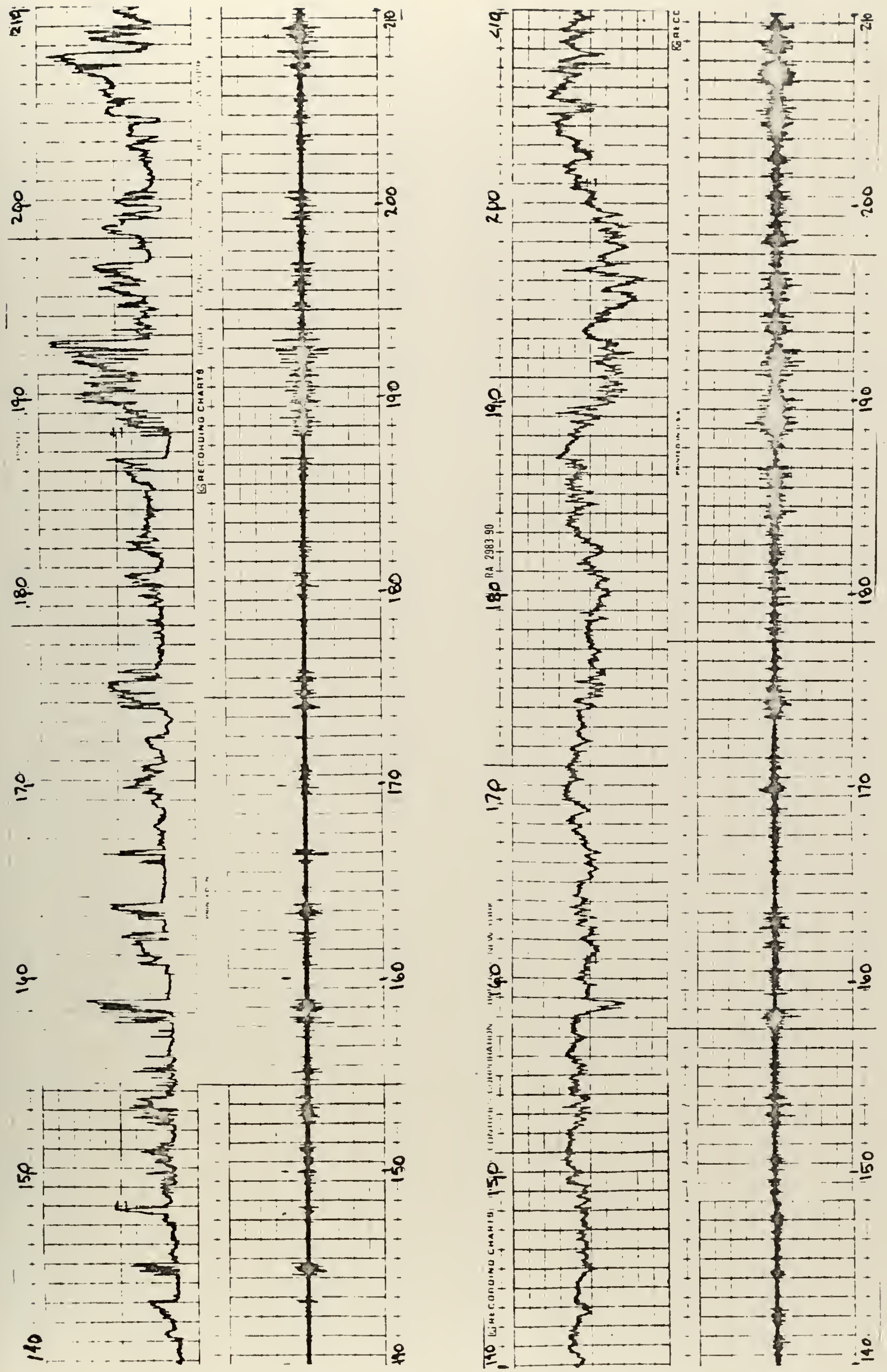
INTEGRAL (SUM) UNDER SPECTRUM = 3.77E 02 N **2
ZEROTH HARMONIC (DC) HAD
AVERAGE = -4.61E 01 N
VARIANCE = 1.09E 03 N **2
TREND = 9.79E-02 N

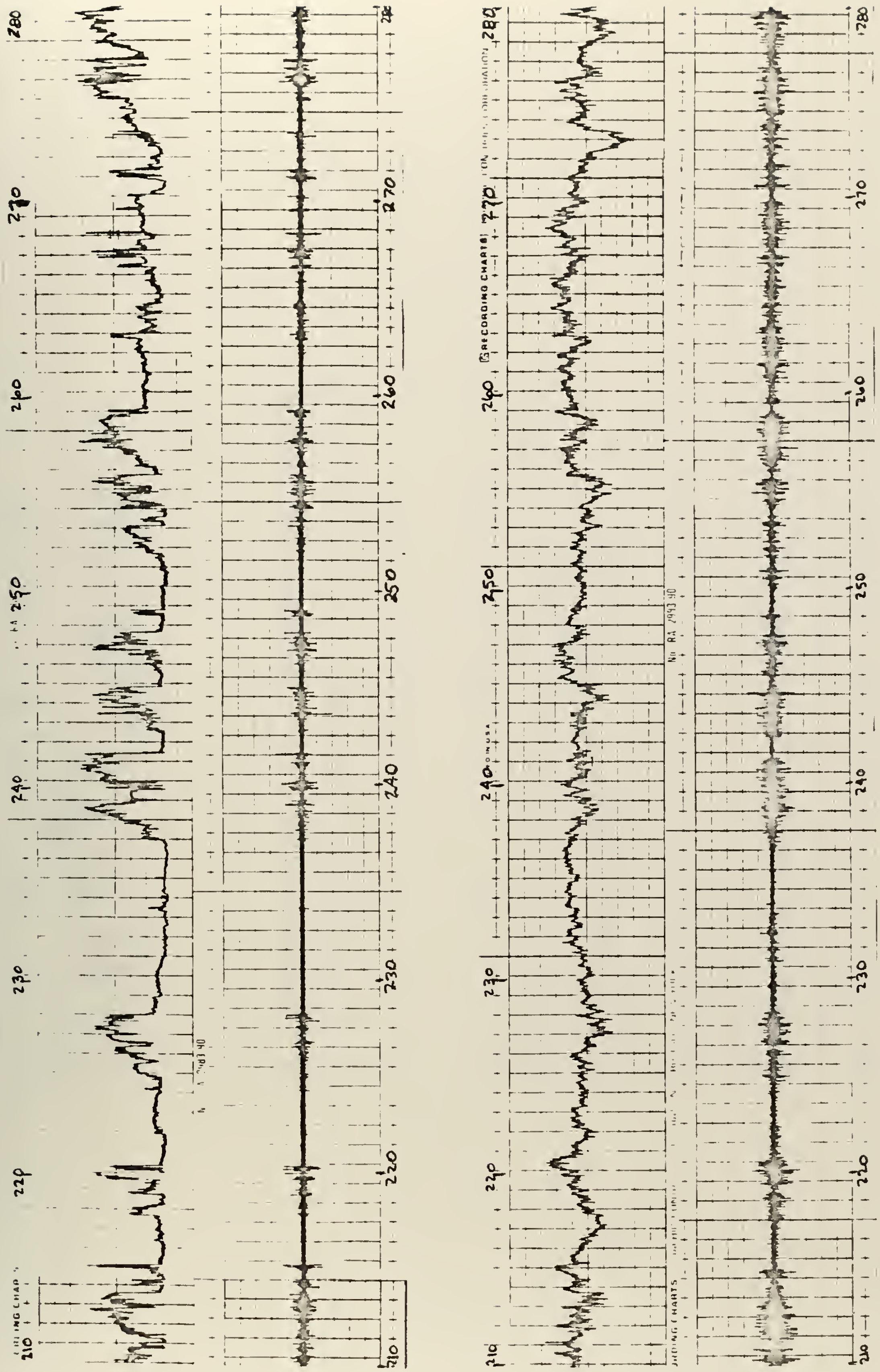
APPENDIX Q. PSD Values for NPS Analysis of Temperature Signal: 306 Seconds of 203(1) U

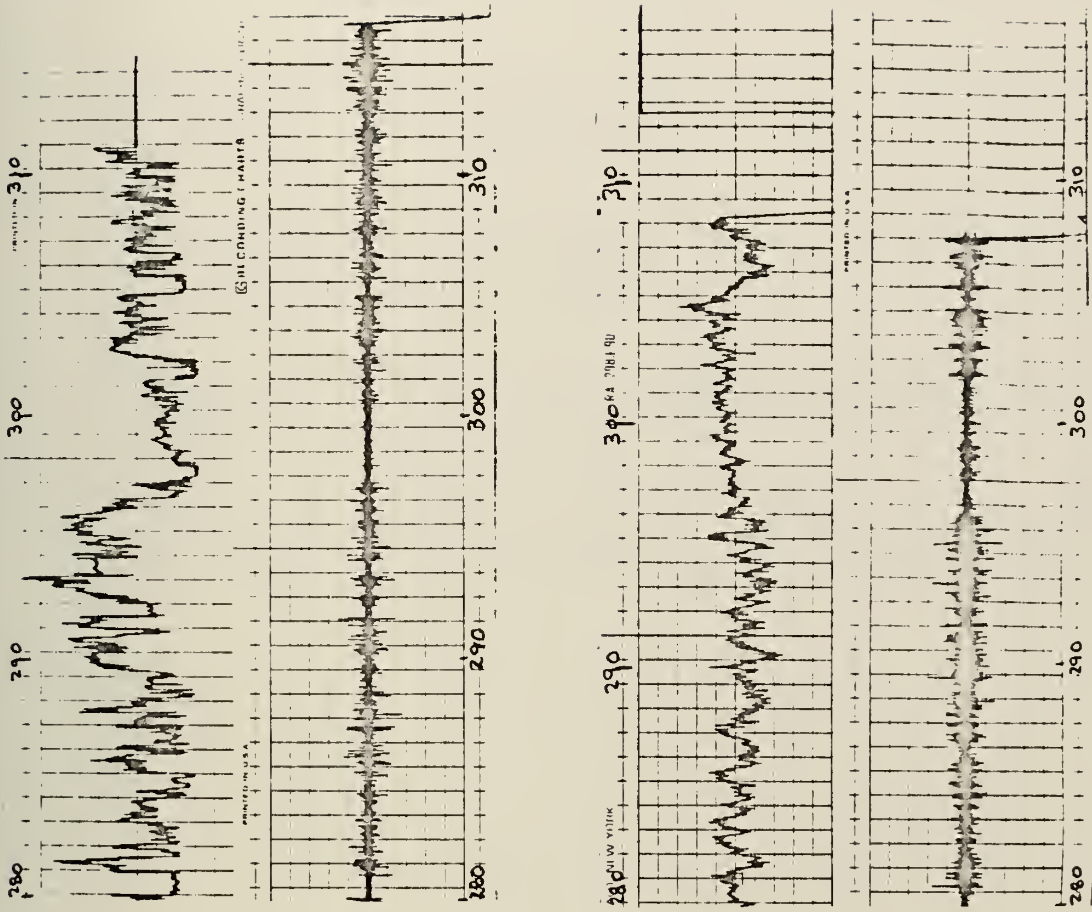


APPENDIX R. Temperature and Velocity Signals: Boston 203(1) El, El', U and U'








280
290
300
310

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13. ABSTRACT

The digitizing procedure used at the Naval Postgraduate School was investigated for possible sources of noise and other errors. Signals of known form were digitized through the Analog-to-Digital Hybrid computer system (Ci 5000/XDS9300). Similar signals were generated by digital programs on the IBM 360/67. The resultant signals were analyzed by the computer programs UBCFTOR, which computed the Fourier coefficients of each block of data, and by UBCSCOR, which computed the power spectra of the signals. The power-spectral plots of the computer-generated signals were compared with the power-spectral plots of digitized signals. The analog-to-digital process appeared to be relatively noise free.

To further test the system, atmospheric temperature and wind velocity signals were digitized and analyzed under UBCFTOR and UBCSCOR. Plots of the time-varying spectra of these signals compared favorably with results obtained at other digitizing facilities.

14. KEY WORDS	LINK A		LINK B		LINK C	
	ROLE	WT	ROLE	WT	ROLE	WT
Turbulence analysis						
Analog-to-Digital Data Conversion						
Digital Data Processing						
Fast-Fourier Transform						
Power Spectral Density						
Spectra Plotting						
Cross Spectral Density						
Time Series Analysis						